

1972

Filter design using minimization techniques.

Maher Ahmed. Sid-Ahmed
University of Windsor

Follow this and additional works at: <http://scholar.uwindsor.ca/etd>

Recommended Citation

Sid-Ahmed, Maher Ahmed., "Filter design using minimization techniques." (1972). *Electronic Theses and Dissertations*. Paper 2105.

This online database contains the full-text of PhD dissertations and Masters' theses of University of Windsor students from 1954 forward. These documents are made available for personal study and research purposes only, in accordance with the Canadian Copyright Act and the Creative Commons license—CC BY-NC-ND (Attribution, Non-Commercial, No Derivative Works). Under this license, works must always be attributed to the copyright holder (original author), cannot be used for any commercial purposes, and may not be altered. Any other use would require the permission of the copyright holder. Students may inquire about withdrawing their dissertation and/or thesis from this database. For additional inquiries, please contact the repository administrator via email (scholarship@uwindsor.ca) or by telephone at 519-253-3000ext. 3208.

FILTER DESIGN USING MINIMIZATION

TECHNIQUES

by

Maher Ahmed Sid Ahmed

A Thesis

Submitted to the Faculty of Graduate Studies through the
Department of Electrical Engineering in Partial Fulfillment
of the Requirements for the Degree of
Master of Applied Science at the
University of Windsor

Windsor , Ontario

1971

© Maher Ahmed Sid Ahmed 1972

380066

ABSTRACT

This work develops methods for choosing the coefficients of a filter transfer function, (digital or analog), to meet required magnitude specifications in the frequency domain.

The method developed make use of the unconstrained minimization technique of Fletcher and Powell⁽⁵⁾, to minimize a magnitude square-error criterion in the frequency domain.

At the end of this work an extension is made to include phase specifications, as well as magnitude specifications.

ACKNOWLEDGMENTS

The writer wishes to express deep gratitude to his supervisor, Dr.G.Jullien , for his guidance and many helpful suggestions during the course of this study.

Thanks are due to Dr.M.Shridhar for his concern and valuable comments.

The financial support provided by the National Research Council of Canada is gratefully acknowledged.

TABLE OF CONTENTS

ABSTRACT	iii
ACKNOWLEDGMENTS	iv
TABLE OF CONTENTS	v
I. INTRODUCTION	1
II. AN OUTLINE OF FLETCHER AND POWELL DESCENT METHOD FOR MINIMIZATION	7
III. USE OF THE UNCONSTRAINED MINIMIZATION TECHNIQUE IN THE DESIGN OF RECURSIVE DIGITAL FILTERS	11
IV. THE DESIGN OF ANALOG FILTERS USING MINIMIZATION TECHNIQUES	46
V. DESIGN OF ANALOG FILTERS USING THE BILINEAR TRANSFORM	54
VI. DESIGN OF DIGITAL FILTERS (USING MINIMIZATION) BY IMPLEMENTING CONSTRAINTS ON AN ANALOG FILTER	82
VII. AN OUTLINE OF A THEORY FOR THE DESIGN OF A DIGITAL FILTER THAT WILL MEET A REQUIRED MAGNITUDE AND PHASE RESPONSE	88
CONCLUSION	95
REFERENCES	98
APPENDIX	99
VITA AUCTORIS	131

CHAPTER I

INTRODUCTION

1.1 LINEAR FILTERING

Linear filtering may be performed digitally by special or general-purpose computers as well as continuously by analog computers or RLC active or passive networks.

This work is aimed at designing digital and analog filters by implementing a minimization technique, to minimize a square error criterion between the magnitude of a transfer function and a desired magnitude-frequency response. The technique used for minimization is a conjugate gradient method derived by Fletcher and Powell.

1.2 ON DIGITAL FILTERING

In this work we shall consider certain classes of frequency-selective linear discrete systems, or digital filters, as they are commonly called. Digital filters utilize digital components as the basic element for implementation. The trends toward decreased cost and size of digital components makes digital filtering especially attractive at this time. These trends promise to end the virtual monopoly of analog components for realizing real-time filters.

Since the first requirement for understanding the concept of the digital filter is a familiarity with difference equations, the next section introduces the subject.

1.3 THE DIFFERENCE EQUATION

The general m th-order difference equation is given by

$$y(nT) = \sum_{i=0}^r L_i x(nT-iT) - \sum_{i=1}^m K_i y(nT-iT) \quad (1.1)$$

where, L_i and K_i are constants.

The form (1.1) shows the iterative nature of the difference equation; given the m previous output values $y(nT-T), y(nT-2T), \text{etc.}$, and the $r+1$ most recent values of the input $x(nT), x(nT-T), \text{etc.}$, the new output may be computed.

A mathematical tool for solving the difference equation is the z -transform.

1.4 DEFINITION OF THE z -TRANSFORM

Given a discrete-time signal f , represented by the sequence of numbers $\{f_n\}$, its z -transform $F(z)$ is defined by the power series

$$F(z) = \sum_{n=0}^{\infty} f_n z^{-n} \quad (1.2)$$

where z is a complex variable. The operation of taking the z -transform of a sequence will be denoted by :

$$F(z) = Z(f_n) \quad (1.3)$$

The z -transform is a linear operation, so that

$$Z(af_n + bg_n) = aZ(f_n) + bZ(g_n) \quad (1.4)$$

where a and b are constants.

An important property that can be derived from (1.2) is the z -transform of a delayed sequence that has zero value at negative time :

$$Z(f_{n-k}) = z^{-k} Z(f_n) \quad (1.5)$$

Equation (1.5) may be used to derive the important property

of the z-transform :

$$F(z) = G(z)H(z) \quad (1.6)$$

This is represented in the time domain as the discrete convolution

$$f_n = \sum_{j=0}^{\infty} h_j g_{n-j} \quad (1.7)$$

where $F(z) = Z(f_n)$, $G(z) = Z(g_n)$, and $H(z) = Z(h_n)$.

1.5 SOLUTION OF THE MTH-ORDER DIFFERENCE EQUATION USING z-TRANSFORM

Take the z-transform of Eq.(1.1), and assume that the initial conditions are zero when the iteration begins.

Therefore,

$$Y(z) = \sum_{l=0}^r L_l X(z) z^{-l} - \sum_{l=1}^m K_l Y(z) z^{-l} \quad (1.8)$$

rewriting

$$Y(z) \sum_{l=0}^m K_l z^{-l} = X(z) \sum_{l=0}^r L_l z^{-l} \quad (1.9)$$

where $K_0 = 1$

or

$$Y(z) = X(z) \frac{\sum_{l=0}^r L_l z^{-l}}{\sum_{l=0}^m K_l z^{-l}} = X(z)H(z) \quad (1.10)$$

Thus the ratio of the z-transform of y_n , and the z-transform of x_n is a system function $H(z)$ which is a rational fraction in z^{-1} , and is a function of the constant coefficients in the original difference equation.

From (1.10) we can derive the canonic form of the mth-order difference equation. Defining an intermediate sequence $e(nT)$ with associated z-transform $E(z)$ such that

$$E(z) = \frac{X(z)}{\sum_{l=0}^m K_l z^{-l}} \quad (1.11)$$

$$Y(z) = E(z) \sum_{i=0}^r L_i z^{-i} \quad (1.12)$$

$$\text{From (1.11)} \quad E(z) = X(z) - E(z) \sum_{i=1}^m K_i z^{-i} \quad (1.13)$$

Eq.(1.13) and (1.12) are used to realize the canonic form shown in fig.(1.1).

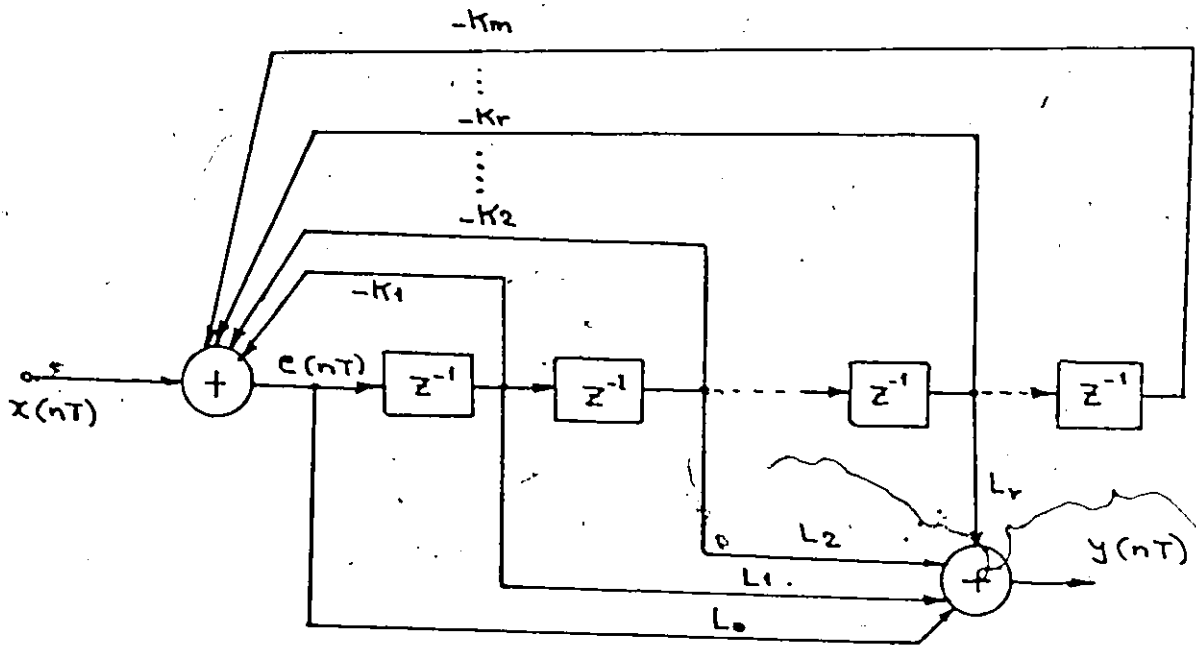


Fig.(1.1) canonic form of the mth-order network.

1.6 INTERPRETATION OF $H(z)$ AS A FREQUENCY SELECTIVE FUNCTION

Assume that the input $x(nT)$ is a sampled complex exponential wave

$$x(nT) = e^{jn\omega T} \quad (1.14)$$

The solution for $y(nT)$ in Eq.(1.1) for such an input can

be written

$$y(nT) = F(e^{j\omega T}) \cdot e^{jn\omega T} \quad (1.15)$$

Therefore, substituting in Eq.(1.1) we get

$$F(e^{j\omega T}) = \frac{\sum_{i=0}^{\infty} L_i e^{-ji\omega T}}{\sum_{i=0}^{\infty} K_i e^{-ji\omega T}} = H(e^{j\omega T}) \quad (1.16)$$

Therefore, when dealing with the frequency domain

$$z = e^{j\omega T} \quad (1.17)$$

1.7 STABILITY OF A DIGITAL FILTER

Consider a Laurent expansion (ref.8) of a function $F(z)$ about some point $z=z_0$.

$$F(z) = \sum_{n=-\infty}^{\infty} a_n (z-z_0)^n \quad (1.18)$$

where

$$a_n = \frac{1}{2\pi j} \oint \frac{F(z)}{(z-z_0)^{n+1}} dz \quad (1.19)$$

From the definition of the z-transform

$$H(z) = \sum_{n=0}^{\infty} h(nT) \cdot z^{-n} \quad (1.20)$$

If we consider this to be a Laurent expansion of the function $H(z)$ about the origin then

$$h(nT) = \frac{1}{2\pi j} \oint \frac{H(z)}{z^{-n+1}} dz \quad (1.21)$$

where c is the contour that encloses all the poles of $H(z)z^{n-1}$ and encircles the origin.

Eq.(1.21) is used at this point to show the significance of pole location on the time response. Consider a filter with a simple pole at $z=a$. Thus we can write

$$H(z) = \frac{H_1(z)}{z - a} \quad (1.22)$$

where $H_1(z)$ is assumed to contain no zeros or poles at $z=a$. By the theorem of residues

$$\frac{1}{2\pi j} \oint \frac{H_1(z)}{z - a} z^{n-1} dz = \sum \text{residues of } \frac{H_1(z)}{z - a} z^{n-1} \quad (1.23)$$

Now, the residue at $z=a$ is given by

$$H_1(a) a^{n-1} \quad (1.24)$$

From Eq.(1.24) the various modes of response as a function of the pole location can be examined. Since, in general, 'a' may be complex, let the pole be described by $a = |a| e^{j\phi}$. Eq.(1.24) then becomes

$$H_1(a) |a|^{n-1} e^{j(n-1)\phi} \quad (1.25)$$

Note that the response time of Eq.(1.24) is an unbounded increasing function of n for $|a| > 1$, and a decaying function for $|a| < 1$.

From this we infer that $H(z)$ is stable if all of its poles lie inside the unit circle.

CHAPTER II

AN OUTLINE OF FLETCHER AND POWELL DESCENT METHOD FOR MINIMIZATION

2.1 INTRODUCTION

A general function $f(X)$ behaves as a pure quadratic function in the vicinity of its minimum. This statement is proved by considering a Taylor series expansion about the minimum X^* . Since the first partial derivatives of $f(X)$ vanish at the minimum X^* , therefore

$$f(X) = f(X^*) + \frac{1}{2}(X-X^*)' A_f(X^*)(X-X^*) \quad (2.1)$$

where $A_f(X)$, the matrix of second partial derivatives of $f(X)$ evaluated at X^* , is positive definite.

Hence, it follows that the only methods which minimize a general function quickly and efficiently are those which (1) work well on a quadratic function and (2) are guaranteed to converge eventually for a general function.

2.2 CONJUGATE DIRECTIONS

A set of n independent and non-zero directions, s_0, s_1, \dots, s_{n-1} are said to be conjugate if, given a symmetric positive definite matrix A , they satisfy

$$s_i' A s_j = 0 \quad 0 \leq i+j \leq n \quad (2.2)$$

and, of course

$$s_1' A s_1 \neq 0.$$

When using conjugate directions in minimizing routines a quadratic function of n variables can be minimized in n -steps.

The general quadratic function can be written

$$q(X) = a + b'AX + \frac{1}{2}X'AX \quad (2.3)$$

where A is positive definite and symmetric. Let X^* minimize $q(X)$. Then

$$\nabla q(X^*) = b + AX^* = 0 \quad (2.4)$$

Given a point X_0 and a set of linearly independent directions $(s_0, s_1, \dots, s_{n-1})$, constants B_i can be found such that

$$X^* = X_0 + \sum_{i=0}^{n-1} B_i s_i \quad (2.5)$$

If the directions s_i are A -conjugate, and none are zero, then B_i can be determined from (2.5) as follows:

$$s_j'AX^* = s_j'AX_0 + \sum_{i=0}^{n-1} B_i s_j'As_i \quad (2.6)$$

$$\text{Using (2.2), } s_j'AX^* = s_j'AX_0 + B_j s_j'As_j$$

and, using (2.4)

$$B_j = -(b + AX_0)' \frac{s_j}{s_j'As_j} \quad (2.7)$$

Now consider an iterative minimization procedure, starting at X_0 and successively minimizing $q(X)$ down the directions s_0, s_1, \dots, s_{n-1} , where these directions satisfy (2.2). Successive points are then determined by the relation

$$X_{i+1} = X_i + \alpha_i s_i \quad i=0,1,\dots,n-1 \quad (2.8)$$

where 'i' is the i^{th} iteration, and α_i is determined by minimizing $f(X_i + \alpha_i s_i)$, so that

$$s_i' \nabla q(X_{i+1}) = 0 \quad (2.9)$$

Using (2.4) in (2.9) gives

$$s_i' (b + A(X_i + \alpha_i s_i)) = 0 \quad (2.10)$$

or,

$$\alpha_i = - (b + AX_i)' \frac{s_i}{s_i' A s_i} \quad (2.11)$$

From (2.8),

$$X_i = X_0 + \sum_{j=0}^{i-1} \alpha_j s_j \quad (2.12)$$

so that

$$X_i' A s_i = X_0' A s_i + \sum_{j=0}^{i-1} \alpha_j s_j' A s_i = X_0' A s_i \quad (2.13)$$

Thus (2.11) becomes

$$\alpha_i = - (b + AX_0)' \frac{s_i}{s_i' A s_i} \quad (2.14)$$

which is identical to (2.7). Hence this sequential process leads,

in n steps, to the minimum X^* .

2.3 METHOD OF FLETCHER AND POWELL

This method is designed so that a quadratic function of n variables is minimized in n iterations.

The method is summarized below:

Assume the function to be minimized is $f(X)$, where X is the vector of the variables.

- 1- Choose an initial point X_0 .
- 2- Form an identity matrix H_0 .
- 3- Calculate the direction of search s_1 from :

$$s_1 = -H \nabla f(X_1)$$

- 4- Choose $\alpha = \alpha_1$ by minimizing $f(X_1 + \alpha s_1)$ with respect to α .
- 5- Calculate the step-length $\sigma_1 = \alpha_1 \cdot s_1$.
- 6- Calculate $X_{1+1} = X_1 + \sigma_1$.
- 7- Update the matrix H_1

$$H_{1+1} = H_1 + A_1 + B_1$$

where

$$A_1 = \frac{\sigma_1 \sigma_1'}{\sigma_1' y_1}, \quad y_1 = \nabla f(X_{1+1}) - \nabla f(X_1)$$

$$B_1 = \frac{H_1 y_1 y_1' H_1}{y_1' H_1 y_1}$$

- 8- Set $i = i+1$ and return to 3.

CHAPTER III

USE OF THE UNCONSTRAINED MINIMIZATION TECHNIQUE IN THE DESIGN OF RECURSIVE DIGITAL FILTERS.

3.1 INTRODUCTION

The purpose of this chapter is to describe a practical method, (which improves on the method proposed by Steiglitz, see sec.3.10), for choosing the coefficients of a recursive digital filter to meet arbitrary specifications of magnitude-frequency response.

The proposed method uses the optimization algorithm described by Fletcher and Powell to minimize a square-error criterion in the frequency domain.

In chapter I, it was proved that the general form of a recursive filter has the form

$$Y(z) = \frac{\sum_{i=0}^r L_i z^{-i}}{1 + \sum_{i=0}^m K_i z^{-i}} \quad (3.1)$$

However, with this form no control can be exercised over the pole location in the z-plane. To remedy this disadvantage a cascade form is chosen.

$$Y(z) = A \prod_{k=1}^K \frac{(z - r_k e^{j\phi_k})(z - r_k e^{-j\phi_k})}{(z - p_k e^{j\theta_k})(z - p_k e^{-j\theta_k})} \prod_{j=1}^L \frac{z - a_j}{z - b_j} \quad (3.2)$$

which could be written in the form

$$Y(z) = A \prod_{k=1}^K \frac{z^2 - 2r_k \cos \phi_k z + r_k^2}{z^2 - 2p_k \cos \theta_k z + p_k^2} \prod_{j=1}^L \frac{z - a_j}{z - b_j} \quad (3.3)$$

3.3 CHOICE OF THE CRITERION FUNCTION

Let the desired magnitude be described at a discrete set of frequencies $\omega_1, \omega_2, \dots, \omega_M$, and let Y_1^d be the desired magnitude at frequency ω_1 .

Choose a square-error criterion

$$Q(\psi_1) = \sum_{i=1}^M (|Y(z_1, \psi_1)| - Y_1^d)^2 \quad (3.4)$$

where $\psi_1 = (r_k, \phi_k, p_k, \theta_k; a_j, b_j; A)^T$ is the $4K+2L+1$ vector of unknown coefficients, and

$$z_1 = e^{j\omega_1 \pi} \quad (3.5)$$

where ω_1 is given in fractions of the Nyquist rate.

3.4 ELIMINATION OF A AS AN UNKNOWN PARAMETER

Steiglitz ref.(1), has shown that it is possible to eliminate A , and hence decrease the number of unknown parameters.

To eliminate A from Q , define the $4K+2L$ dimensional vector

$$\Psi = (r_k, \phi_k, p_k, \theta_k; a_j, b_j)^T \quad (3.6)$$

and write

$$Y(z, A, \Psi) = A \prod_{k=1}^K \frac{z^2 - (2r_k \cos \phi_k)z + r_k^2}{z^2 - (2p_k \cos \theta_k)z + p_k^2} \prod_{j=1}^L \frac{z - a_j}{z - b_j} = AH(z, \Psi) \quad (3.7)$$

Then

$$Q(A, \Psi) = \sum_{i=1}^M (|AH(z_i, \Psi)| - y_i^d)^2 \quad (3.8)$$

Differentiate (3.8) with respect to A and set the result to zero to obtain the optimum value of A . Hence A^* (optimum A) is given by

$$|A^*| = \frac{\sum_{i=1}^M |H(z_i, \Psi)| y_i^d}{\sum_{i=1}^M |H(z_i, \Psi)|^2} \quad (3.9)$$

Since the sign of A^* does not affect the magnitude characteristic, it will be taken as positive.

The Fletcher-Powell method is used to minimize the new error-criterion

$$Q(\Psi) = Q(A^*, \Psi) \quad (3.10)$$

3.5 CALCULATION OF THE GRADIENT OF Q WITH RESPECT TO ψ

The Fletcher-Powell method requires the calculation of the gradient of Q with respect to the parameter vector ψ .

Since,
$$Q = \sum_{i=1}^M (A^* |H(z_i, \psi)| - y_i^d)^2$$

Therefore,

$$\frac{\partial Q}{\partial \psi} = 2A^* \sum_{i=1}^M (A^* |H(z_i, \psi)| - y_i^d) \frac{\partial |H(z_i, \psi)|}{\partial \psi} \quad (3.11)$$

Writing

$$|H(z_i, \psi)| = (H(z_i, \psi) \overline{H(z_i, \psi)})^{\frac{1}{2}}$$

We have

$$\frac{\partial |H(z_i, \psi)|}{\partial \psi} = \frac{1}{H(z_i, \psi)} \cdot \text{Re}(\overline{H(z_i, \psi)}) \frac{\partial H(z_i, \psi)}{\partial \psi} \quad (3.12)$$

3.6 STABILITY CONSIDERATIONS

For stability the poles should lie inside the unit circle

of the z -plane , and to insure minimum phase constraint the zeros shall be constrained to lie also in the unit circle of the z -plane.

Therefore,

$$\begin{aligned} 0 &\leq r_k \leq 1 \\ 0 &\leq p_k \leq 1 \\ -1 &\leq a_j \leq 1 \\ -1 &\leq b_j \leq 1 \end{aligned} \quad (3.13)$$

To secure the above constraints , we choose a new set of variables , ref.(7), related to the above as follows

$$\begin{aligned} r_k &= \sin^2 \alpha_k \\ p_k &= \sin^2 \beta_k \\ a_j &= \sin \theta_{zj} \\ b_j &= \sin \theta_{pj} \end{aligned} \quad (3.14)$$

Therefore,

$$H(z, \psi) = \prod_{k=1}^K \frac{z^2 - (2\sin^2 \alpha_k \cdot \cos \phi_k)z + \sin^4 \alpha_k}{z^2 - (2\sin^2 \beta_k \cdot \cos \theta_k)z + \sin^4 \beta_k} \prod_{j=1}^L \frac{z - \sin \theta_{zj}}{z - \sin \theta_{pj}} \quad (3.15)$$

3.7 CALCULATION OF $\frac{\partial |H(z_1, \psi)|}{\partial \psi}$

Since;

$$\frac{\partial |H(z_1, \psi)|}{\partial \psi} = \frac{1}{|H(z_1, \psi)|} \operatorname{Re}(H(z_1, \psi) \frac{\partial H(z_1, \psi)}{\partial \psi})$$

Therefore;

$$\frac{\partial |H(z_1, \psi)|}{\partial \alpha_k} = |H(z_1, \psi)| \cdot \operatorname{Re} \left(\frac{-4 \sin \alpha_k \cdot \cos \alpha_k \cdot \cos \phi_k \cdot z_1 + 4 \sin^3 \alpha_k \cdot \cos \alpha_k}{z_1^2 - (2 \sin^2 \alpha_k \cdot \cos \phi_k) z_1 + \sin^4 \alpha_k} \right)$$

$$\frac{\partial |H(z_1, \psi)|}{\partial \phi_k} = |H(z_1, \psi)| \cdot \operatorname{Re} \left(\frac{2 \sin^2 \alpha_k \cdot \sin \phi_k \cdot z_1}{z_1^2 - (2 \sin^2 \alpha_k \cdot \cos \phi_k) z_1 + \sin^4 \alpha_k} \right)$$

$$\frac{\partial |H(z_1, \psi)|}{\partial \beta_k} = -|H(z_1, \psi)| \cdot \operatorname{Re} \left(\frac{-4 \sin \beta_k \cdot \cos \beta_k \cdot \cos \theta_k \cdot z_1 + 4 \sin^3 \beta_k \cdot \cos \beta_k}{z_1^2 - (2 \sin^2 \beta_k \cdot \cos \theta_k) z_1 + \sin^4 \beta_k} \right)$$

$$\frac{\partial H(z_1, \psi)}{\partial \theta_k} = -|H(z_1, \psi)| \cdot \operatorname{Re} \left(\frac{2 \sin^2 \theta_k \cdot \sin \theta_k \cdot z_1}{z_1^2 - (2 \sin^2 \theta_k \cdot \cos \theta_k) z_1 + \sin^4 \theta_k} \right)$$

$$\frac{\partial |H(z_1, \psi)|}{\partial \theta_{zj}} = -|H(z_1, \psi)| \cdot \operatorname{Re} \left(\frac{\cos \theta_{zj}}{z_1 - \sin \theta_{zj}} \right)$$

$$\frac{\partial |H(z_1, \psi)|}{\partial \theta_{pj}} = -|H(z_1, \psi)| \cdot \operatorname{Re} \left(\frac{\cos \theta_{pj}}{z_1 - \sin \theta_{pj}} \right)$$

In order to use Fletcher-Powell subroutine (FMFP or DFMFP) supplied by IEM in the scientific subroutine package, another subroutine FUNCT(N,X,Q,DQ) is written in which N is the number of variables, X is the vector of the variables, Q is the criterion function, and DQ is the gradient of the criterion function with respect to the parameter vector X. This subroutine is summarized in the next section.

3.8 SUMMARY OF THE SUBROUTINE FUNCT(N,X,Q,DQ)

1) Calculate H_i given by eq.(3.15) $i=1,2,\dots,M$

2) Calculate A^* given by eq.(3.9)

3) Calculate

$$E_i = A^* |H_i| - Y_i^d \quad i=1,2,\dots,M$$

4) Calculate

$$Q = \sum_{i=1}^M E_i^2$$

5) Calculate

$$\frac{\partial |H_i|}{\partial \psi_n}, \quad n=1,2,\dots,4K+2L$$

(Formulas are given in previous section)

6) Calculate

$$\frac{\partial Q}{\partial \psi_n} = 2A^* \sum_{i=1}^M E_i \frac{\partial |H_i|}{\partial \psi_n}, \quad n=1,2,\dots,4K+2L$$

This subroutine was written in double precision complex arithmetic in Fortran IV

3.9 EXAMPLES

Example 1 : A low-pass filter

Specification of the filter are

$$w=0.0, 0.09(0.01); \quad Y^d=1.0$$

i.e. w is taken in steps of 0.01 from 0.0 to 0.09

$$w=0.10; \quad Y^d=0.5$$

$$w=0.11, 0.20(0.01); \quad Y^d=0.0$$

$$w=0.2, 1.0(0.1); \quad Y^d=0.0$$

If optimization was started for $K=1, L=1$ at

$$\Psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$$

After 10 iterations

$$Q=0.3659$$

$$A^*=0.6926$$

$$\text{Poles } \begin{pmatrix} 0.9092 \pm j0.2330 \\ 0.8735 \end{pmatrix}$$

$$\text{Zeros } \begin{pmatrix} 0.06557 \pm j0.03398 \\ -0.3188 \end{pmatrix}$$

If we assume the filter to have the following form

$$Y(z) = A \frac{z^{-2} + a_1 z^{-1} + b_1}{z^{-2} + c_1 z^{-1} + d_1} \frac{1 - a_2 z^{-1}}{1 - b_2 z^{-1}}$$

Then,

$$a_1 = -0.13114$$

$$b_1 = 0.54550$$

$$c_1 = -0.18183$$

$$d_1 = 0.8809$$

$$a_{z1} = -0.31877$$

$$b_{p1} = 0.87348$$

If now optimization is started for $K=2$, $L=1$, and ψ_0 the same as before, then after 30 iterations

$$Q = 0.09887$$

$$A^* = 0.5449$$

$$\begin{aligned} & (0.90097 \pm j0.24487) \\ \text{Poles } & (0.89287 \pm j0.28659) \\ & (0.87316) \end{aligned}$$

$$\begin{aligned} & (0.93166 \pm j0.35903) \\ \text{Zeros } & (0.601387 \pm j0.75623) \\ & (-0.6666) \end{aligned}$$

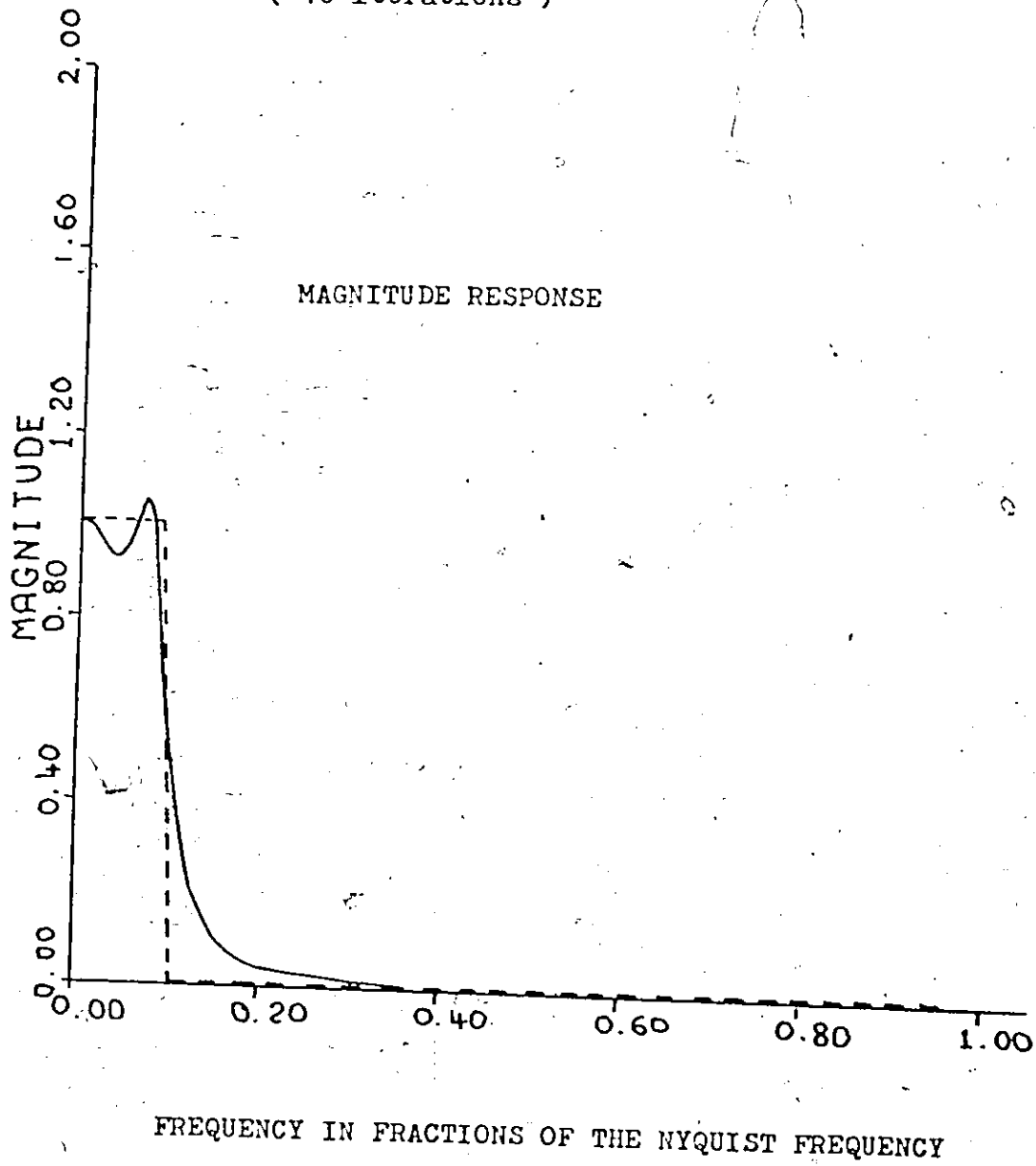
and if the transfer function is in the form

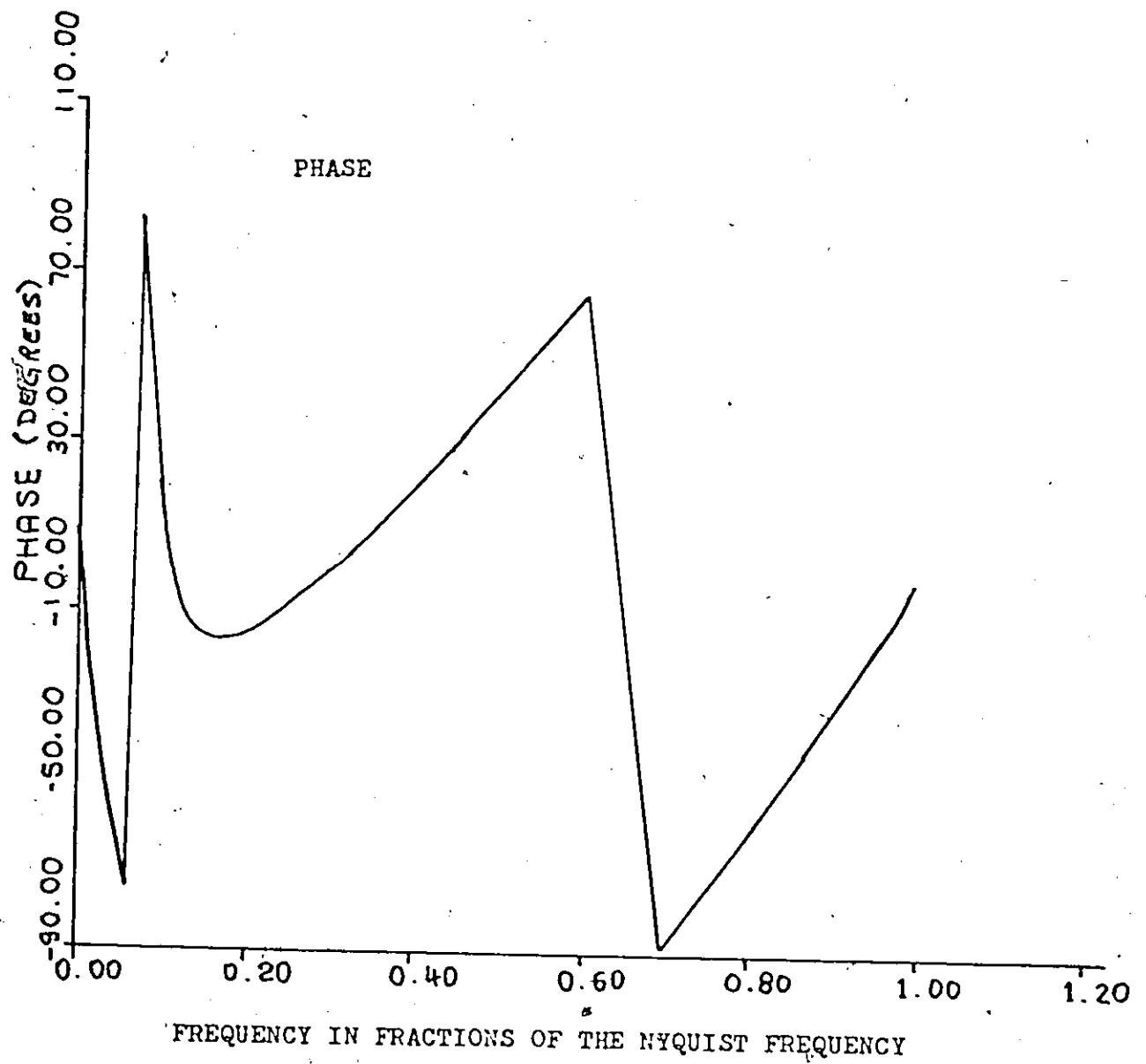
$$Y(z_1) = A \cdot \frac{1+a_1 z^{-1}+b_1 z^{-2}}{1+c_1 z^{-1}+d_1 z^{-2}} \cdot \frac{1+a_2 z^{-1}+b_2 z^{-2}}{1+c_2 z^{-1}+d_2 z^{-2}} \cdot \frac{1-a_{z1} z^{-1}}{1-b_{p1} z^{-1}}$$

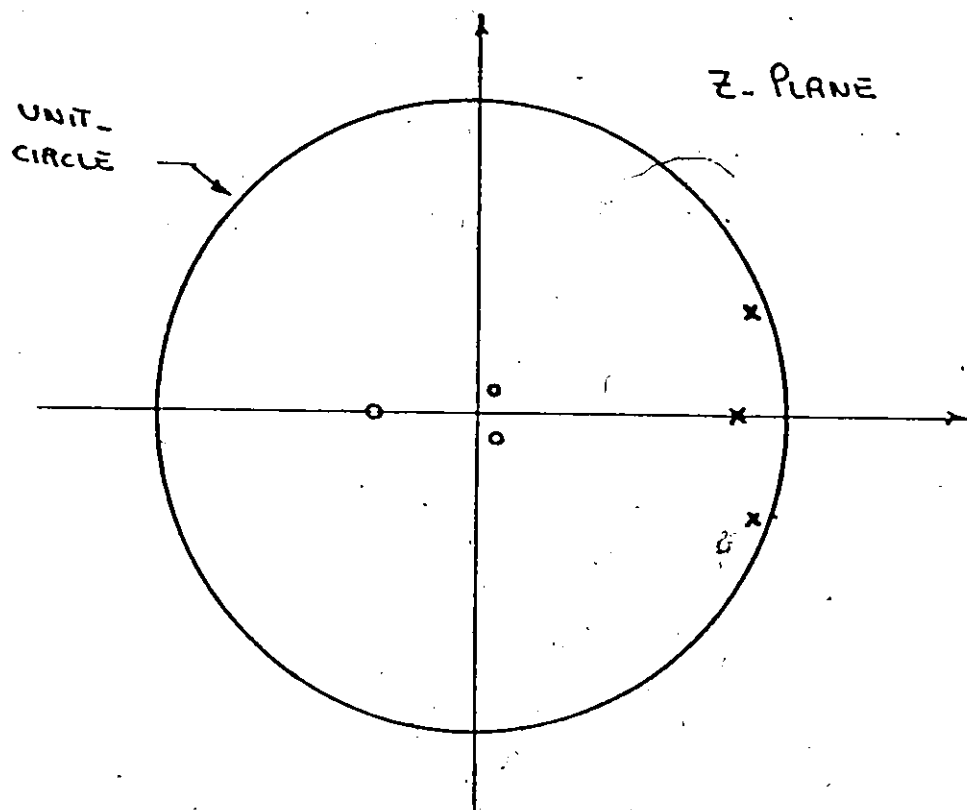
Example 1: Third order recursive digital filter

$$Q = 0.3659$$

(10 iterations)







POLE-ZERO LOCATION OF EXAMPLE 1 (THIRD-ORDER)

Then,

$$a_1 = -1.86333$$

$$b_1 = 0.99689$$

$$c_1 = -1.80196$$

$$d_1 = 0.87173$$

$$a_2 = -1.20277$$

$$b_2 = 0.93355$$

$$c_2 = -1.78574$$

$$d_2 = 0.87935$$

$$a_{z1} = -0.6666$$

$$b_{p1} = 0.87316$$

Example 2 : Design a high pass recursive digital filter having the following specifications

$$w = 0.0, 0.495(0.05); \quad Y^d = 0.0$$

$$w = 0.5; \quad Y^d = 0.5$$

$$w = 0.505, 1(0.05); \quad Y^d = 1.0$$

$K=1, L=1$ was considered, and ψ_0 was chosen to be

$$\psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$$

After 30 iterations the results of the filter were

$$Q = 0.5969$$

$$A^* = 0.3016$$

$$\text{Poles } (0.1487 \pm j0.1357) \\ (0.2506)$$

$$\begin{array}{l} \text{Zeros } (0.60866 \pm j0.78877) \\ (0.91841) \end{array}$$

$$a_1 = -1.21732$$

$$b_1 = 0.99263$$

$$c_1 = -0.29748$$

$$d_1 = 0.0405510$$

$$a_{z1} = 0.91841$$

$$b_{p1} = 0.25068$$

If, now, an eighth order high-pass filter was designed taking $K=3$, $L=2$, and ψ_0 was taken as the final ψ of the previous case.

$$\begin{array}{l} \psi_0 = (1.63, 0.914, -0.465, 0.740; (\text{repeated } 3 \text{ times, since } K=3); \\ 1.64, 0.2534; (\text{repeated } 2 \text{ times, since } L=2)). \end{array}$$

After 17 iterations

$$Q = 0.33179$$

$$A^* = 0.06659$$

$$a_1 = -0.10853 \times 10$$

$$a_2 = a_3 = a_1$$

$$b_1 = 0.99316$$

$$b_2 = b_3 = b_1$$

$$c_1 = -0.13864$$

$$c_2 = c_3 = c_1$$

$$d_1 = 0.28580$$

$$d_2 = d_3 = d_1$$

$$\text{Poles } 1 = (0.69321 \times 10^{-1} \pm j0.53009)$$

=complex poles of the first cascade

=Poles 2 =Poles 3.

$$\text{Zeros } 1 = (0.54266 \pm j0.83587)$$

=complex zeros of the first cascade

=Zeros 2 =Zeros 3.

$$a_{z1} = 0.91782$$

$$a_{z2} = a_{z1}$$

$$b_{p1} = 0.51083$$

$$b_{p2} = b_{p1}$$

After 30 iterations

$$Q = 0.091689$$

$$A^* = 0.05905$$

$$a_1 = -1.01448$$

$$b_1 = 0.86266$$

$$c_1 = -0.14435$$

$$d_1 = 0.328583$$

$$a_2 = -1.06575$$

$$b_2 = 0.92916$$

$$c_2=0.06223$$

$$d_2=0.46850$$

$$a_3=-1.03250$$

$$b_3=0.89096$$

$$c_3=0.01237$$

$$d_3=0.34901$$

$$a_{z1}=0.99286$$

$$b_{p1}=0.52533$$

$$a_{z2}=0.99761$$

$$b_{p2}=0.48769$$

$$(0.72175 \times 10^{-1} \pm j0.56866)$$

$$\text{Poles } (-0.311163 \times 10^{-1} \pm j0.6837)$$

$$(-0.61889 \times 10^{-2} \pm j0.59074)$$

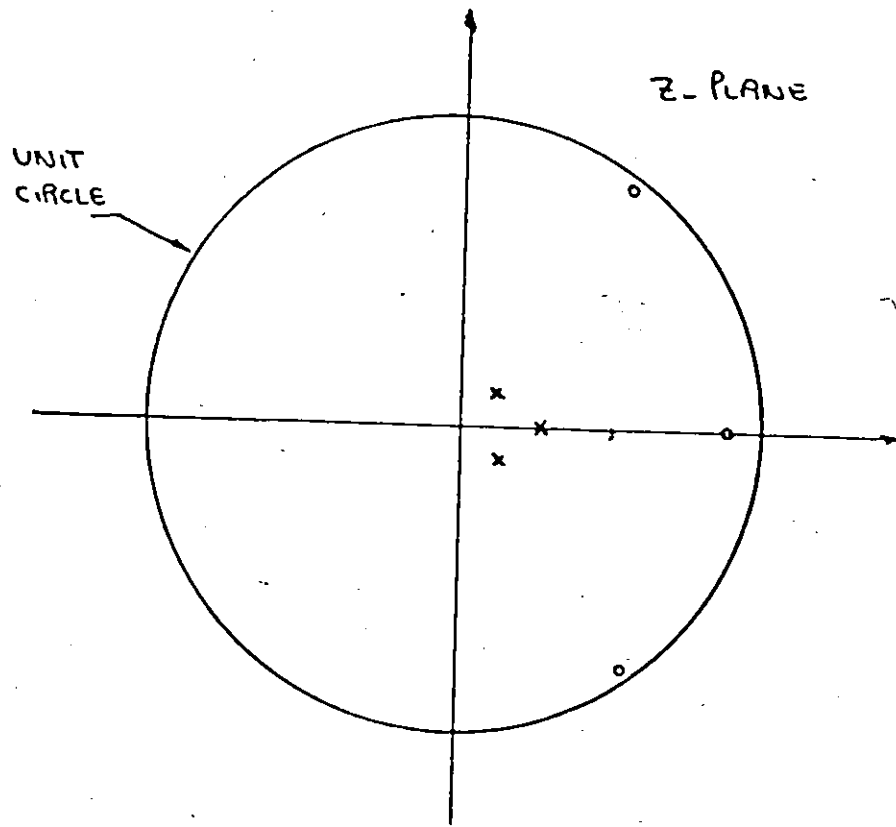
$$(b_{p1}, \text{ and } b_{p2})$$

$$(0.507240 \pm j0.778058)$$

$$\text{Zeros } (0.532874 \pm j0.803245)$$

$$(0.516253 \pm j0.790220)$$

$$(a_{z1}, \text{ and } a_{z2})$$

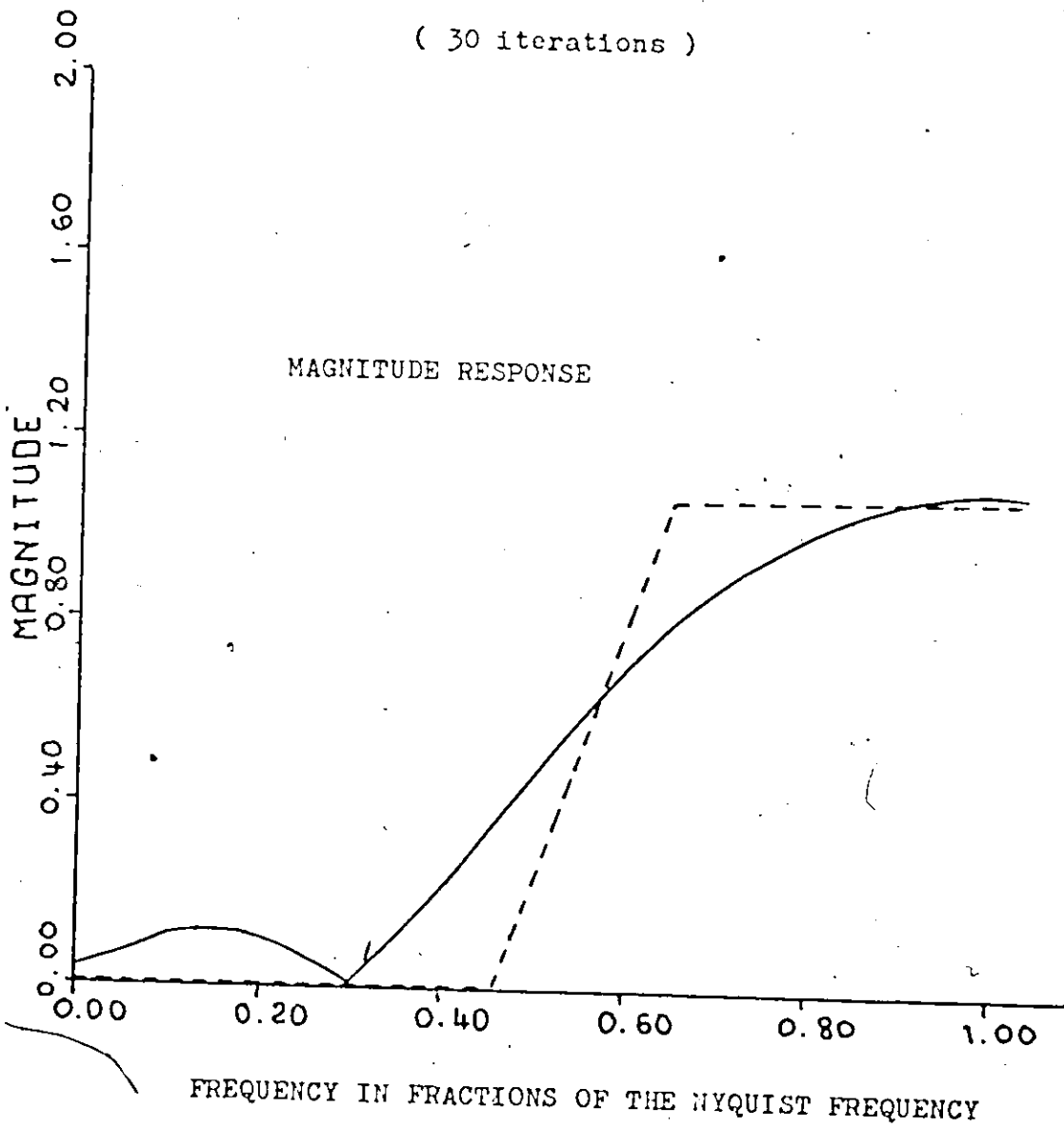


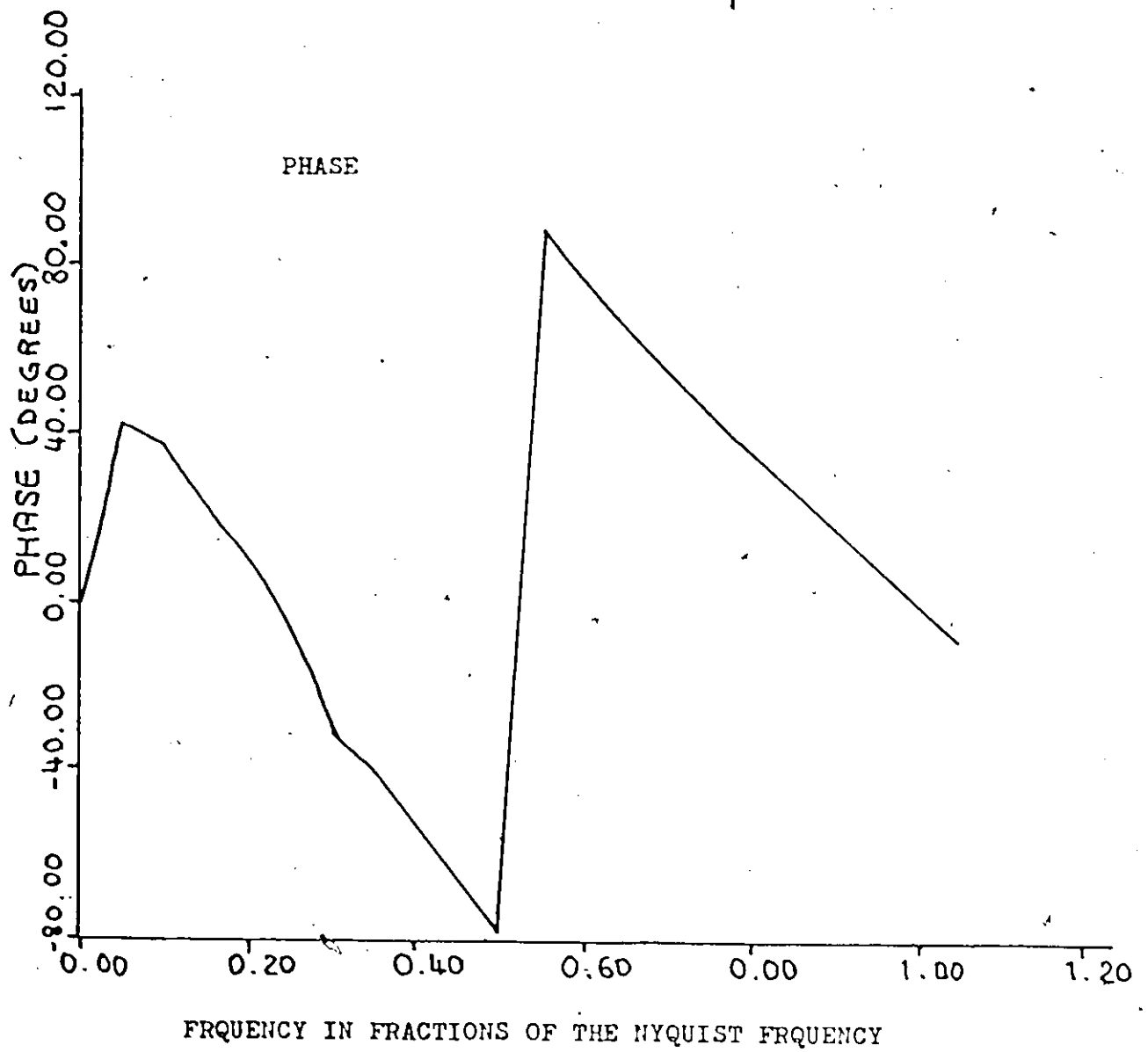
2 POLE-ZERO LOCATION OF EXAMPLE 2 (THIRD ORDER)

Example 2 : Third order digital high-pass filter

$$Q = 0.5969$$

(30 iterations)

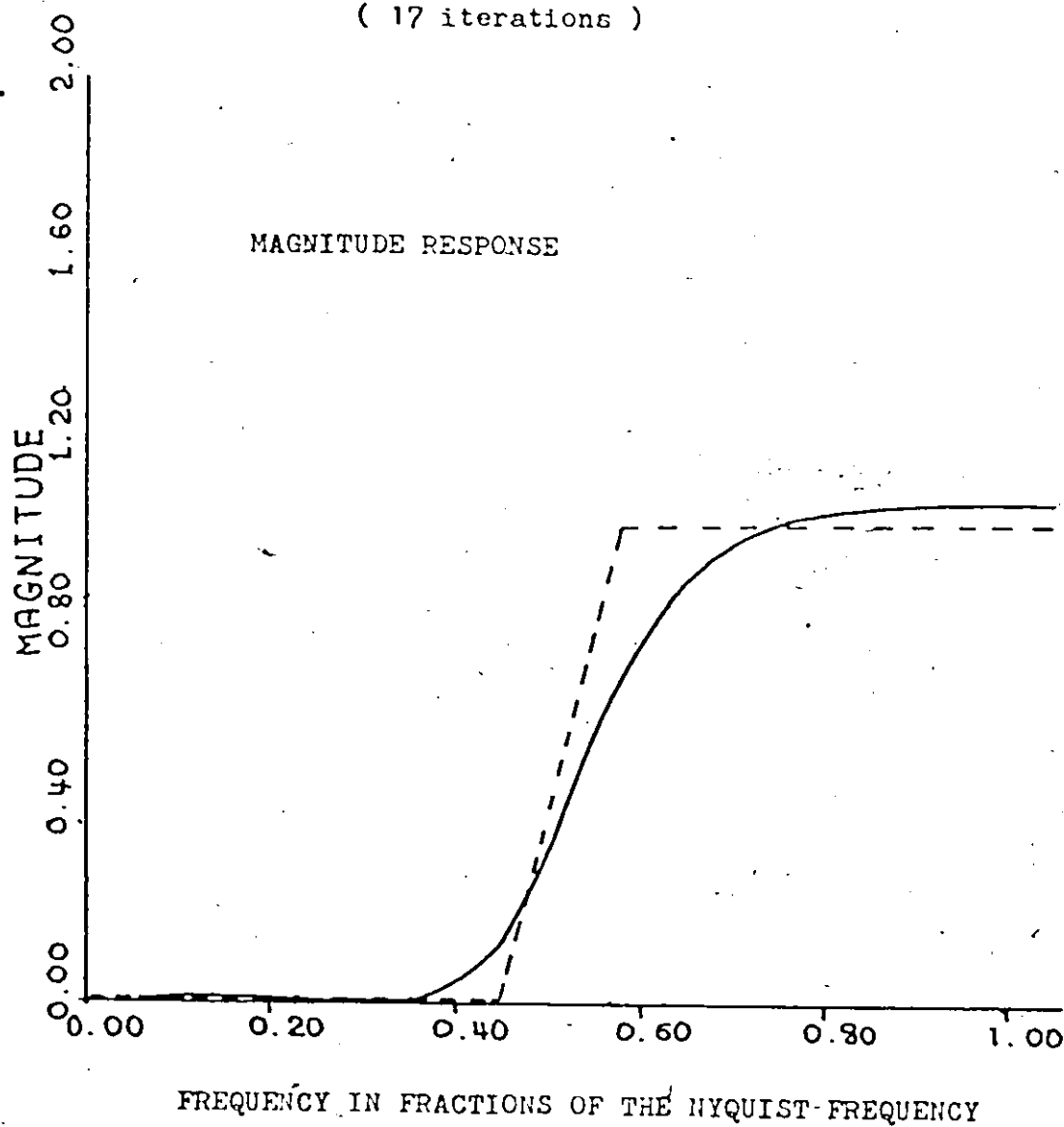


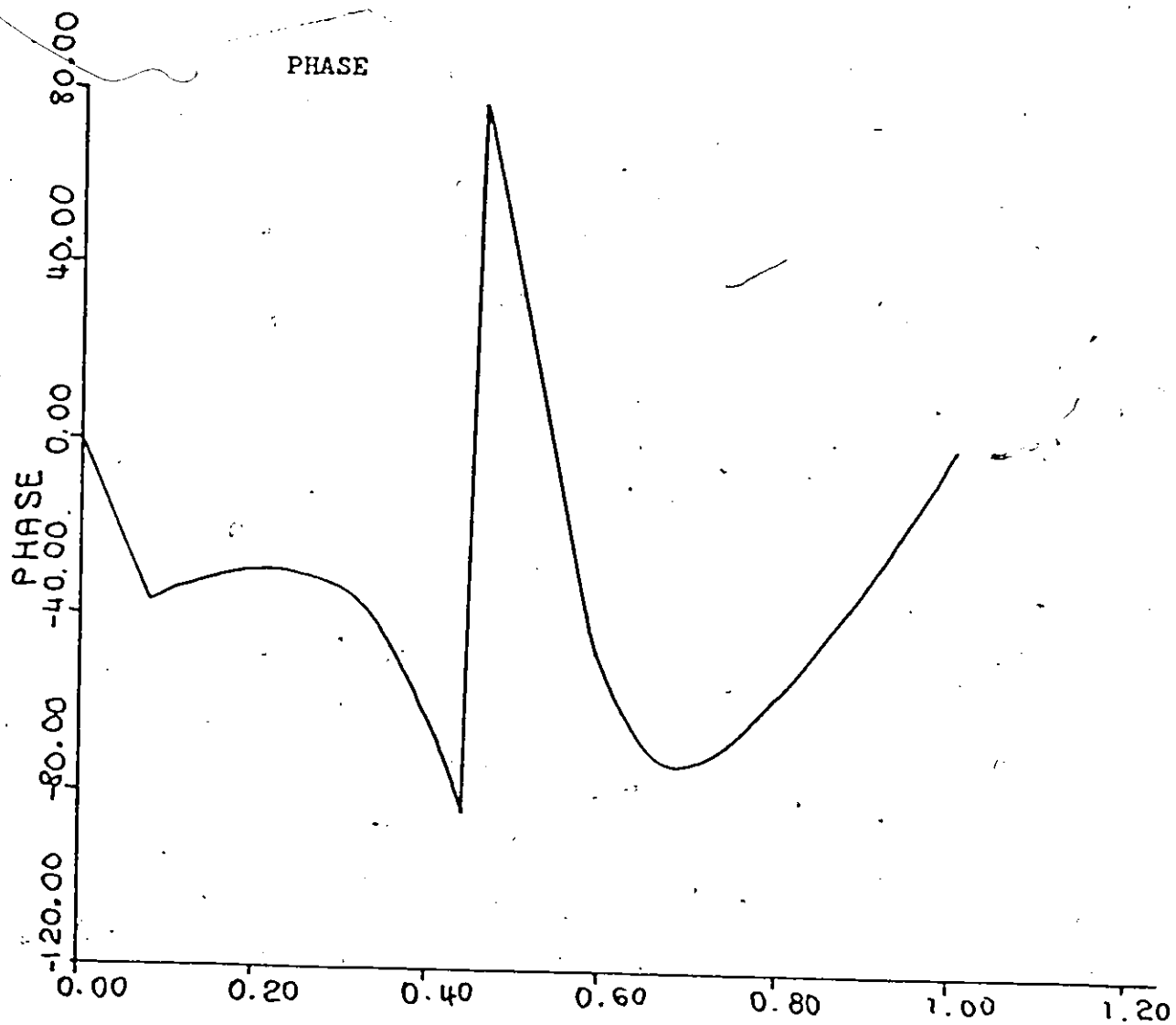


Example 2 : Eighth order digital high-pass filter

$$Q = 0.3318$$

(17 iterations)



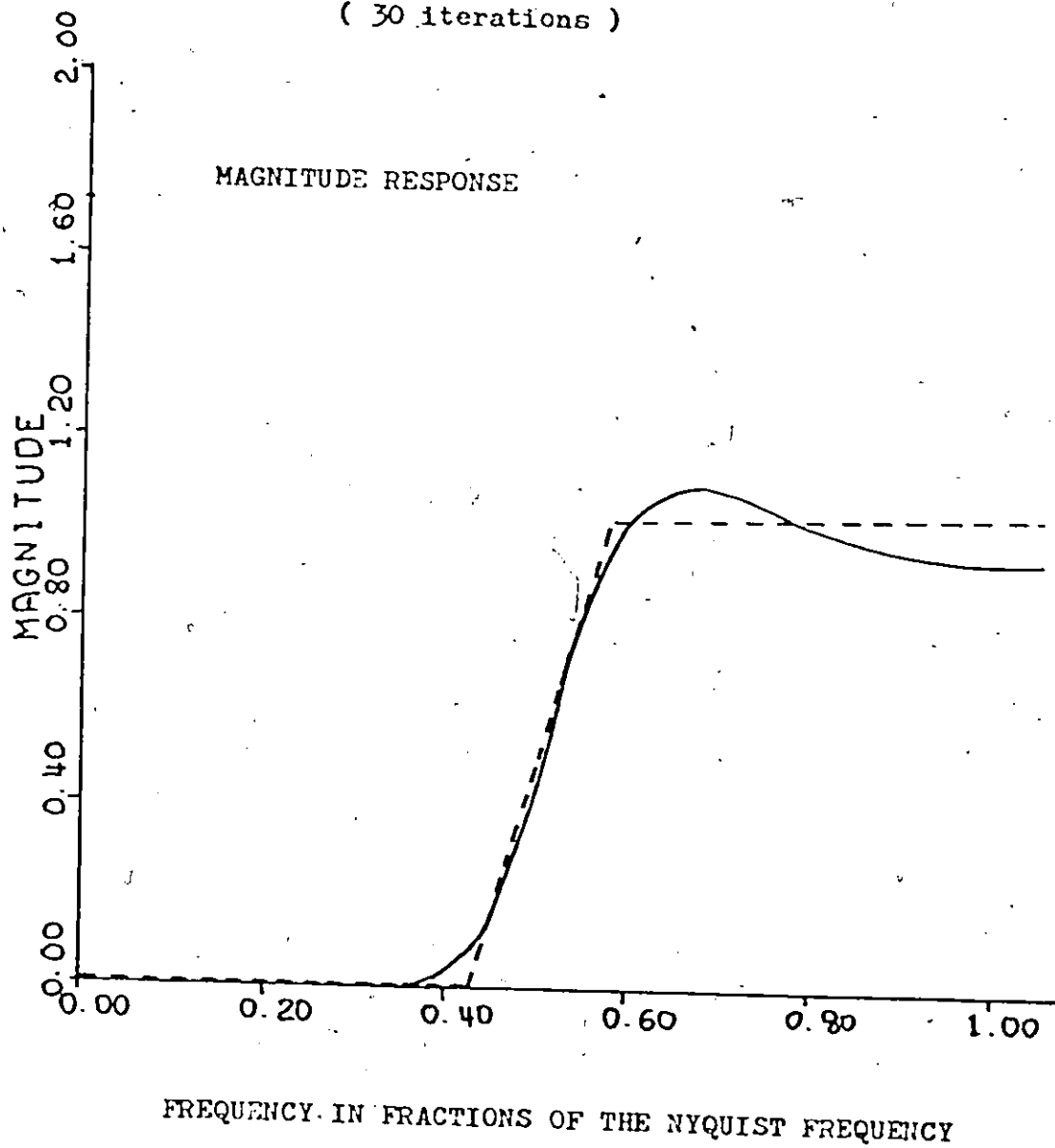


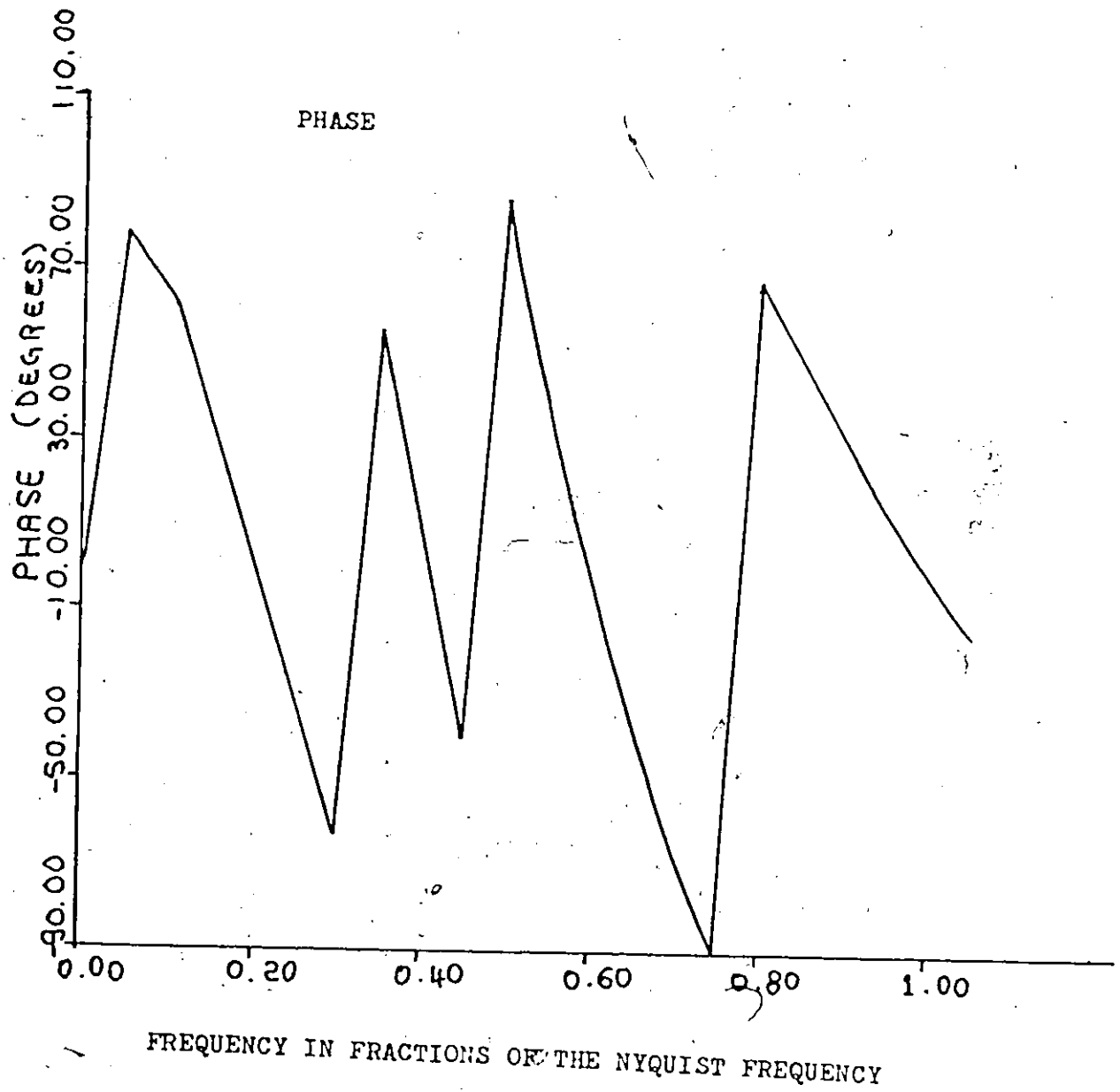
FREQUENCY IN FRACTIONS OF THE NYQUIST FREQUENCY

Example 2 : Eighth order digital high-pass filter.

$$Q = 0.0916898$$

(30 iterations)





Example 3 : Design a recursive digital band-pass filter having the following specifications

$$\begin{array}{ll} w=0.0, 0.32(0.08); & Y^d=0.0 \\ w=0.4; & Y^d=0.5 \\ w=0.402, 0.508(0.02); & Y^d=1.0 \\ w=0.60; & Y^d=0.5 \\ w=0.604, 1.0(0.04); & Y^d=0.0 \end{array}$$

Starting with $\Psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$, and taking $K=L=1$, after 30 iterations

$$Q=1.3190$$

$$A^*=0.2412$$

$$a_1 = -0.4483 \times 10^{-3}$$

$$b_1 = 0.8809 \times 10^{-7}$$

$$c_1 = 0.0106365$$

$$d_1 = 0.74572$$

$$a_{z1} = 0.93338$$

$$b_{p1} = 0.49479$$

$$\begin{array}{l} \text{Poles } (-0.531824 \times 10^{-2} \pm j0.863534) \\ (b_{p1}) \end{array}$$

$$\begin{array}{l} \text{Zeros } (0.22415 \times 10^{-3} \pm j0.194518 \times 10^{-3}) \\ (a_{z1}) \end{array}$$

If , now an eighth order band-pass filter was designed taking $K=3$, $L=2$; and Ψ_0 was taken as the final Ψ of the previous case, i.e. $\Psi_0 = (0.017228, -0.71479, 1.19244, 1.5769; \text{ (repeated 3 times, since } K=3 \text{)}; 1.93787, 0.517595; \text{ (repeated 2 times, since } L=2 \text{)})$. After 30 iterations

$$Q = 0.642177$$

$$A^* = 0.104296$$

$$a_1 = -0.169412 \times 10^{-3}$$

$$b_1 = 0.125339 \times 10^{-7}$$

$$c_1 = -0.61250 \times 10^{-1}$$

$$d_1 = 0.475182$$

$$a_2 = -0.16941 \times 10^{-3}$$

$$b_2 = b_1$$

$$c_2 = 0.250454$$

$$d_2 = 0.617215$$

$$a_3 = -0.16038 \times 10^{-3}$$

$$b_3 = 0.112358 \times 10^{-7}$$

$$c_3 = -0.21611$$

$$d_3 = 0.438632$$

$$a_{z1} = 0.94557$$

$$b_{p1} = 0.48471$$

$$a_{z2} = a_{z1}$$

$$b_{z2} = 0.48172$$

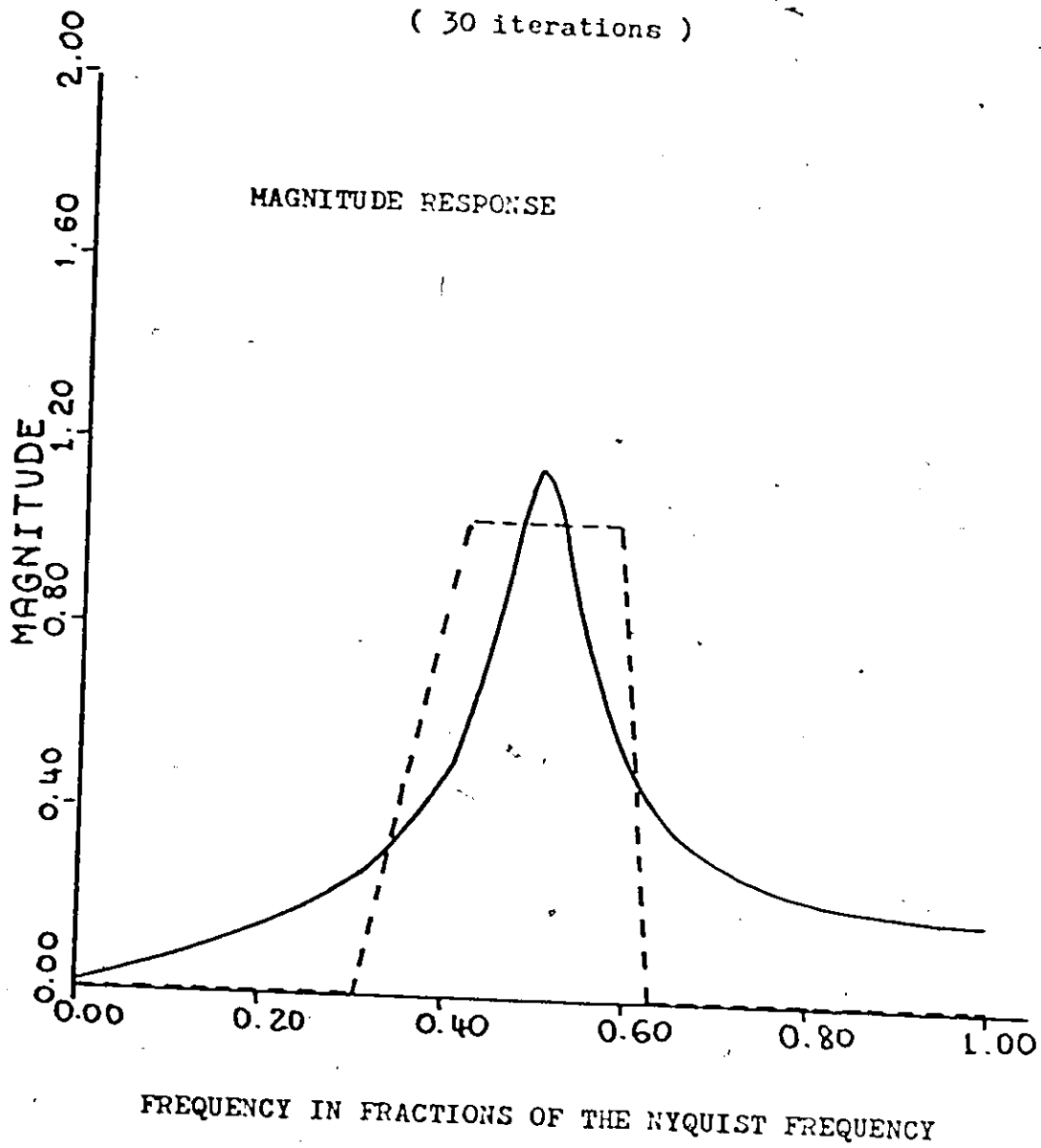
Poles $(0.30625 \times 10^{-1} \pm j0.68865)$
 $(-0.1252 \pm j0.77558)$
 $(0.10805 \pm j0.65342)$
 $(b_{p1} \text{ and } b_{p2})$

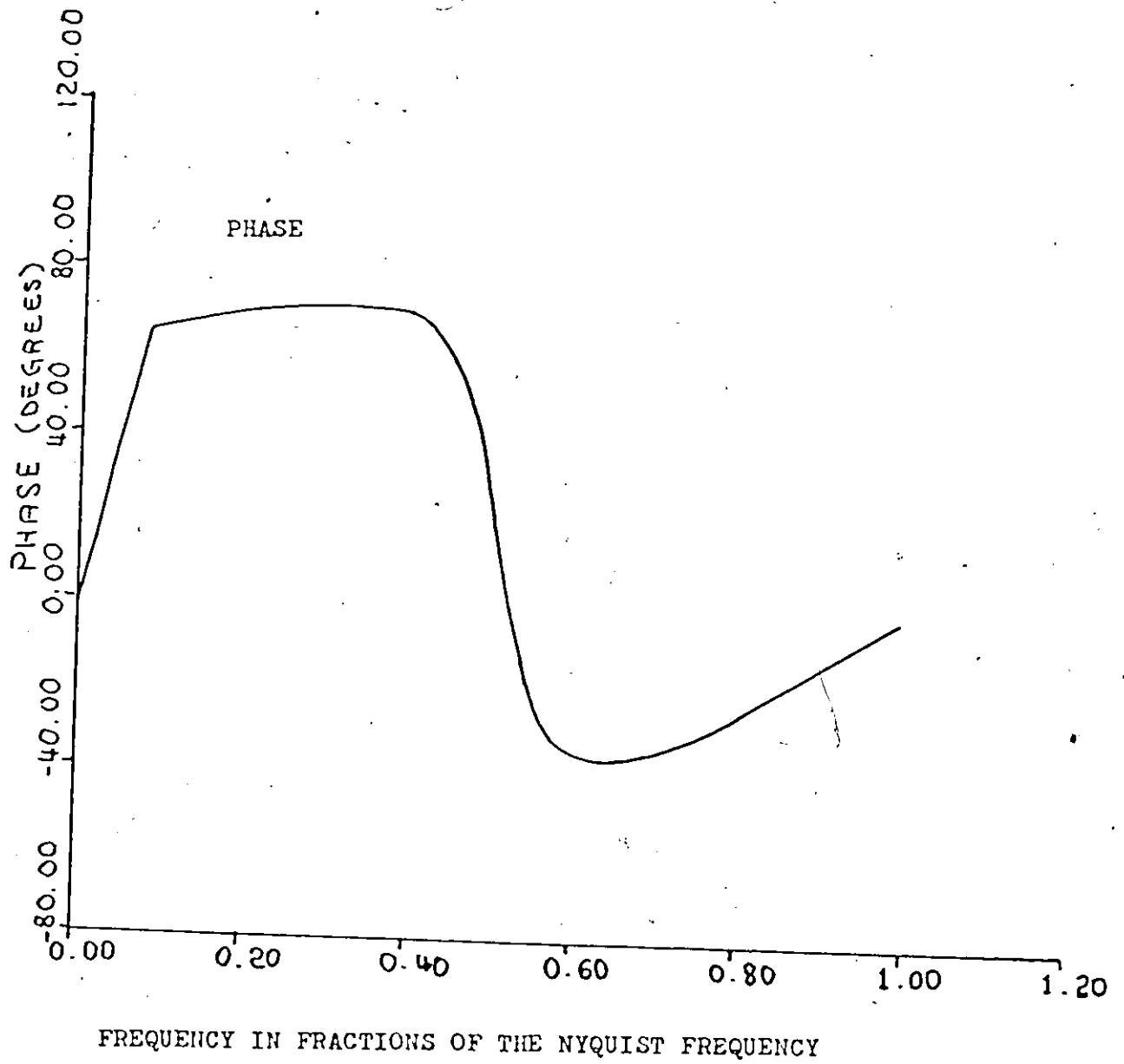
Zeros $(0.847064 \times 10^{-4} \pm j0.73203 \times 10^{-4})$
 $(0.847064 \times 10^{-4} \pm j0.73203 \times 10^{-4})$
 $(0.801906 \times 10^{-4} \pm j0.69320 \times 10^{-4})$
 $(a_{z1} \text{ and } a_{z2})$

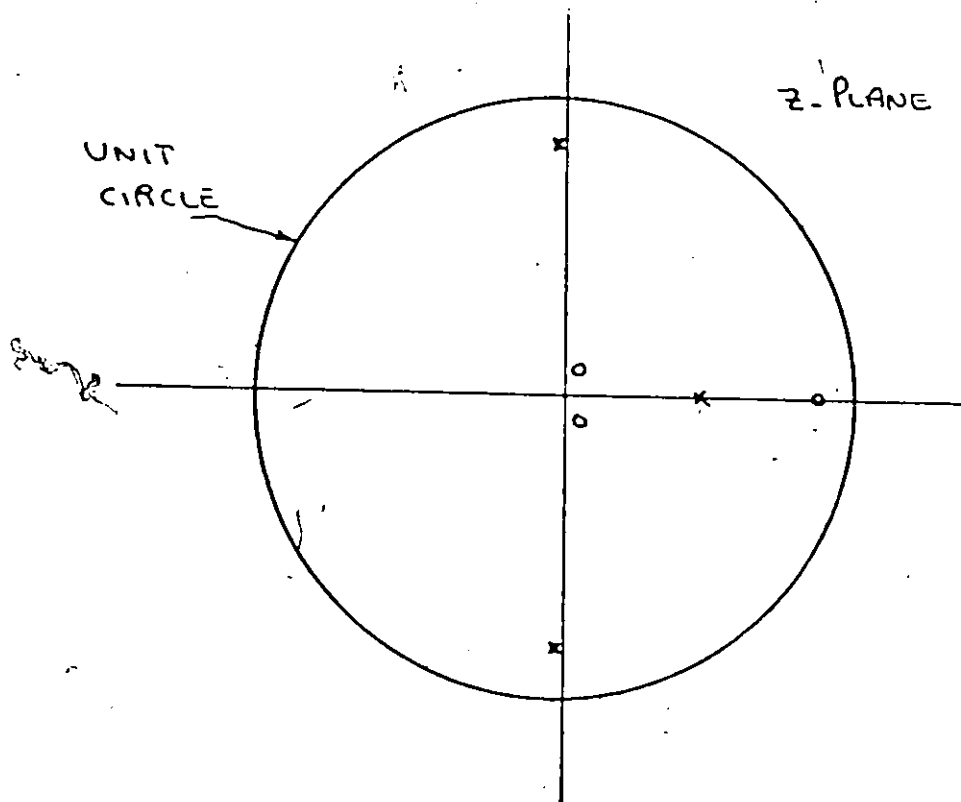
Example 3 : Third order band-pass filter

$$Q = 1.3198$$

(30 iterations)



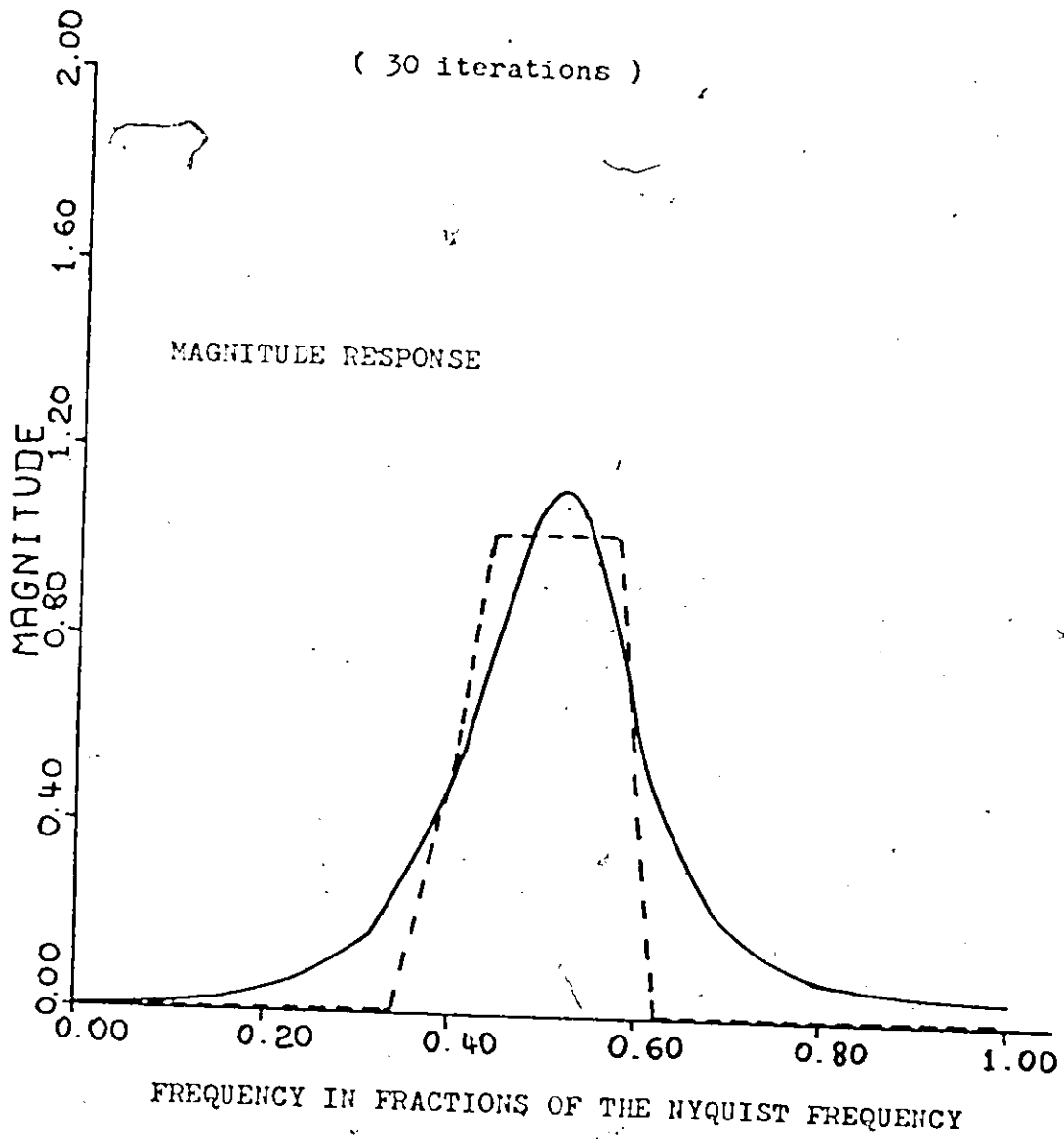


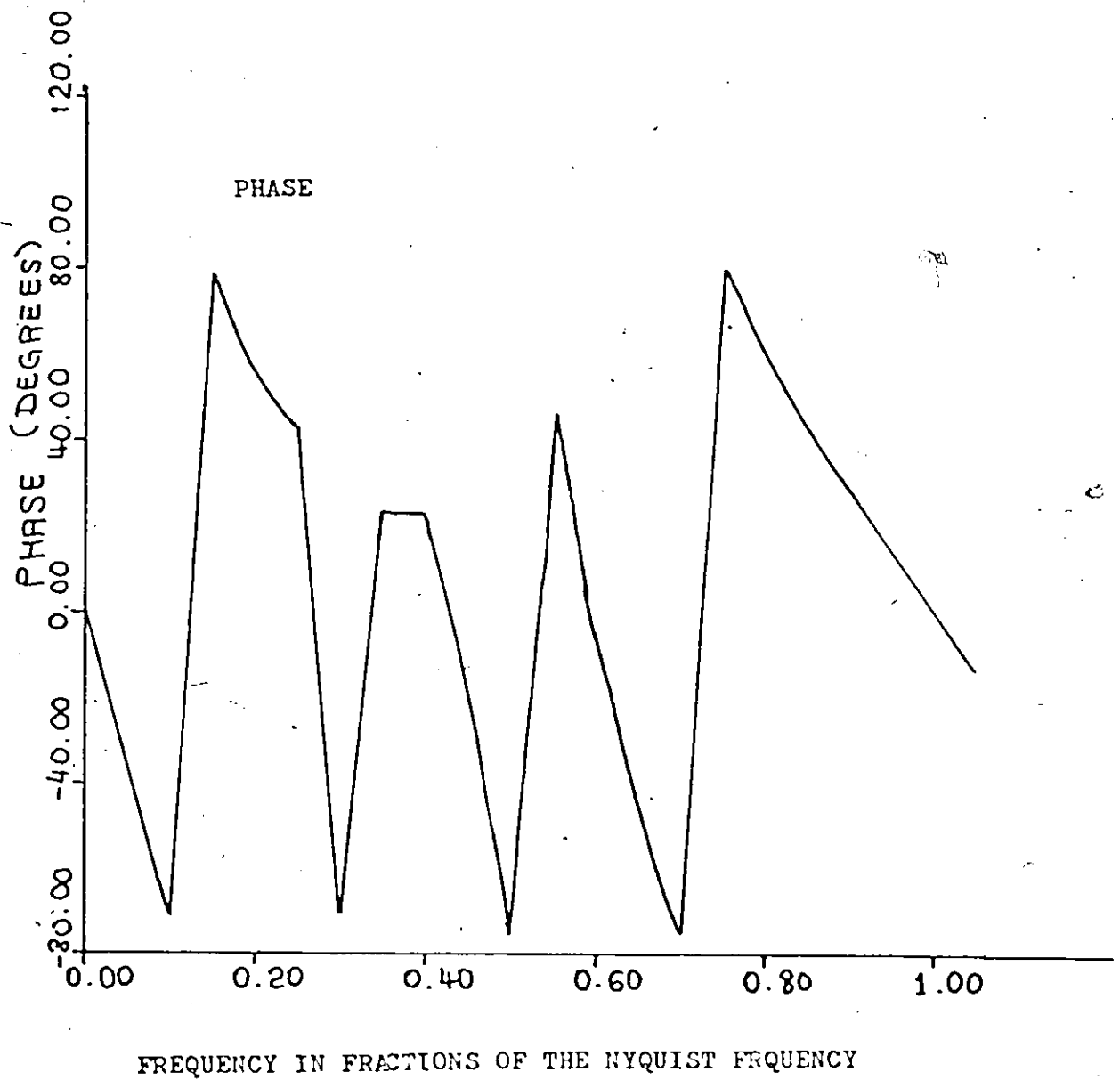


POLE-ZERO LOCATION OF THIRD ORDER BAND-PASS FILTER

Example 3 : eighth-order band-pass digital filter

$$Q = 0.64178$$





3.10 A REVIEW OF STEIGLITZ'S METHOD⁽¹⁾

A technique of minimizing a square-error criterion to obtain a required digital filter transfer function has been reported by Steiglitz⁽¹⁾.

This chapter introduces an improvement on Steiglitz's method of implementing constraints on the poles and zeros.

The method adopted in ref.(1) is as follows :

If $Y(z)$ has a pole or a zero outside the unit circle , then inverting this pole or zero with respect to the unit circle does not affect the magnitude response , since the effect of inverting a pole at $|\alpha|e^{j\theta}$ with respect to the unit circle is a multiplication by the function

$$\frac{z - |\alpha|e^{j\theta}}{z - \frac{1}{|\alpha|}e^{j\theta}}$$

$$\text{Now, } z - |\alpha|e^{j\theta} = (1 - 2|\alpha|\cos(\omega n - \theta) + |\alpha|^2)^{\frac{1}{2}}$$

$$\begin{aligned} z - \frac{1}{|\alpha|}e^{j\theta} &= (1 - 2\frac{1}{|\alpha|}\cos(\omega n - \theta) + \frac{1}{|\alpha|^2})^{\frac{1}{2}} \\ &= \frac{1}{|\alpha|} (1 - 2|\alpha|\cos(\omega n - \theta) + |\alpha|^2)^{\frac{1}{2}} \end{aligned}$$

This is effectively multiplying the magnitude of $Y(z)$ by $|\alpha|$, since A is chosen optimally this does not affect the magnitude response.

The steps given in ref.(1) are as follows :

- 1) Start from a transfer function

$$Y(z,X) = A \prod_{k=1}^K \frac{1+a_k z^{-1}+b_k z^{-2}}{1+c_k z^{-1}+d_k z^{-2}}$$

where, $X=(a_k, b_k, c_k, d_k)^T$

- 2) Minimize the square error criterion given by eq.(3.8) .
- 3) At the optimum, (or a specified number of iterations), invert the poles or zeros lying outside the unit circle.

It has also been found that when starting anew from the result of step 3, further reduction in Q is possible.

The principal disadvantage of this procedure is the necessity of a search in the infinite z -plane .

It is possible to carry out this search in the unit circle where the poles and zeros are required to be located.

The method adopted in this chapter permits determination of a small value of Q in a relatively small number of iterations as compared with the method proposed by Steiglitz⁽¹⁾, irrespective of the initial pole-zero location.

In example 1, which was taken from ref.(1), Q reached a value of 0.3659 in 10 iterations using the method proposed in this chapter,

while it took 93 iterations to reach a value of $Q=1.2611$ (for $K=1$), and a value of $Q=0.56731$ in 155 iterations, using Steiglitz's method.

CHAPTER IV

THE DESIGN OF ANALOG FILTERS USING MINIMIZATION TECHNIQUE

4.1 INTRODUCTION

The purpose of this chapter is to describe a method for choosing the coefficients of an analog filter to meet arbitrary specifications of magnitude response.

The method described does not design a filter for the whole infinite frequency range, but only a specified finite range of frequency.

The proposed method uses the optimization algorithm described by Fletcher and Powell to minimize a square-error criterion in the frequency domain.

4.2 THE FILTER FORM

To control the pole and zero locations of the analog filter, the following cascade form is chosen

$$G(s) = A \prod_{k=1}^K \frac{(s - j_k^2 e^{j\phi_k})(s - j_k^2 e^{j\phi_k})}{(s - p_k^2 e^{j\theta_k})(s - p_k^2 e^{j\theta_k})} \prod_{j=1}^L \frac{(s + \sigma_{zj}^2)}{(s + \sigma_{pj}^2)} \quad (4.1)$$

which can be written in the form

$$G(s) = A \prod_{k=1}^K \frac{s^2 - (2f_k^2 \cos \alpha_k) s + f_k^4}{s^2 - (2p_k^2 \cos \theta_k') s + p_k^4} \prod_{j=1}^L \frac{(s + \sigma_{zj}^2)}{(s + \sigma_{pj}^2)} \quad (4.2)$$

4.3 THE CRITERION FUNCTION

Let the desired magnitude be described at a discrete set of frequencies $\omega_1, \omega_2, \dots, \omega_M$, and let Y_1^d be the desired magnitude at the frequency ω_1 .

Choose a square-error criterion

$$Q(\psi_1) = \sum_{i=1}^M (|G(s_i, \psi_1)| - Y_1^d)^2 \quad (4.3)$$

where, $\psi_1 = (f_k, \phi_k, p_k, \theta_k'; \sigma_{zj}, \sigma_{pj}; A)^T$, is the $4K+2L+1$ vector of unknown coefficients, and

$$s_1 = j\omega_1 \quad (4.4)$$

4.4 ELIMINATION OF A AS AN UNKNOWN PARAMETER

To eliminate A from Q, define the $4K+2L$ dimensional parameter vector

$$\psi = (f_k, \phi_k, p_k, \theta_k'; \sigma_{zj}, \sigma_{pj})^T \quad (4.5)$$

and write

$$G(s, A, \psi) = AH(s, \psi) \quad (4.6)$$

Then

$$Q(A, \psi) = \sum_{i=1}^M (|AH(s_i, \psi)| - y_1^d)^2 \quad (4.7)$$

Differentiate (4.7) with respect to A and set the result to zero

$$\frac{\partial Q(A, \psi)}{\partial |A|} = 2 \sum_{i=1}^M (|AH(s_i, \psi)| - y_1^d) |H(s_i, \psi)| = 0.$$

Therefore

$$|A^*| = \frac{\sum_{i=1}^M |H(s_i, \psi)| y_1^d}{\sum_{i=1}^M |H(s_i, \psi)|^2} \quad (4.8)$$

Since the sign of A^* does not affect the magnitude characteristic, it will be taken as positive.

The Fletcher-Powell method is used to minimize the new error-criterion

$$Q(\psi) = Q(A^*, \psi) \quad (4.9)$$

4.5 CALCULATION OF THE GRADIENT OF Q WITH RESPECT TO ψ

The Fletcher-Powell method requires the calculation of Q with respect to the parameter vector ψ .

Equations expressing the gradient of Q with respect to ψ are given in section 3.5 .

4.6 STABILITY CONSIDERATIONS

For the filter to be stable with minimum phase constraint the poles and zeros should lie in the left-half of the s -plane.

Hence,
$$\frac{\pi}{2} \leq \phi_k \leq \pi$$

and using variable substitution

$$\phi_k = \frac{\pi}{2} + \frac{\pi}{2} \sin^2 \alpha_k \quad (4.10)$$

also

$$\frac{\pi}{2} \leq \theta_k \leq \pi$$

using variable substitution

$$\theta_k = \frac{\pi}{2} + \frac{\pi}{2} \sin^2 \theta_k \quad (4.11)$$

where θ_k and α_k are the substitution variables.

Substituting (4.10) and (4.11) we get

$$G(s) = A \prod_{k=1}^K \frac{s^2 + 2p_k^2 \sin(\frac{\pi}{2} \sin^2 \alpha_k) \cdot s + p_k^4}{s^2 + 2p_k^2 \sin(\frac{\pi}{2} \sin^2 \theta_k) \cdot s + p_k^4} \prod_{j=1}^L \frac{s^2 + \sigma_{zj}^2}{s^2 + \sigma_{pj}^2} \quad (4.12)$$

4.7 CALCULATION OF $\frac{\partial |H(s_1, \psi)|}{\partial \psi}$

Since,

$$\frac{\partial |H(s_1, \psi)|}{\partial \psi} = \frac{1}{|H(s_1, \psi)|} \cdot \operatorname{Re} \left(\frac{H(s_1, \psi)}{|H(s_1, \psi)|} \cdot \frac{\partial H(s_1, \psi)}{\partial \psi} \right)$$

Therefore

$$\frac{\partial |H(s_1, \psi)|}{\partial f_k} = |H_1| \cdot \operatorname{Re} \left(\frac{4f_k \cdot \sin\left(\frac{\pi}{2} \sin^2 \alpha_k\right) \cdot s_1 + 4f_k^3}{s_1^2 + 2f_k^2 \cdot \sin\left(\frac{\pi}{2} \sin^2 \alpha_k\right) \cdot s_1 + f_k^4} \right)$$

$$\frac{\partial |H_1|}{\partial \alpha_k} = |H_1| \cdot \operatorname{Re} \left(\frac{2f_k^2 \cdot \cos\left(\frac{\pi}{2} \sin^2 \alpha_k\right) \cdot \frac{\pi}{2} \cdot 2 \sin \alpha_k \cdot \cos \alpha_k \cdot s_1}{s_1^2 + 2f_k^2 \cdot \sin\left(\frac{\pi}{2} \sin^2 \alpha_k\right) \cdot s_1 + f_k^4} \right)$$

$$\frac{\partial |H_1|}{\partial p_k} = |H_1| \cdot \operatorname{Re} \left(\frac{4p_k \cdot \sin\left(\frac{\pi}{2} \sin^2 \theta_k\right) \cdot s_1 + 4p_k^3}{s_1^2 + 2p_k^2 \cdot \sin\left(\frac{\pi}{2} \sin^2 \theta_k\right) \cdot s_1 + p_k^4} \right)$$

$$\frac{\partial |H_1|}{\partial \theta_k} = -|H_1| \cdot \operatorname{Re} \left(\frac{2p_k^2 \cdot \cos\left(\frac{\pi}{2} \sin^2 \theta_k\right) \cdot \pi \cdot \sin \theta_k \cdot \cos \theta_k \cdot s_1}{s_1^2 + 2p_k^2 \cdot \sin\left(\frac{\pi}{2} \sin^2 \theta_k\right) \cdot s_1 + p_k^4} \right)$$

$$\frac{\partial |H_i|}{\partial \sigma_{zj}} = |H_i| \cdot \operatorname{Re} \left(\frac{2\sigma_{zj}}{s + \sigma_{zj}^2} \right)$$

$$\frac{\partial |H_i|}{\partial \sigma_{pj}} = -|H_i| \cdot \operatorname{Re} \left(\frac{2\sigma_{pj}}{s + \sigma_{pj}^2} \right)$$

4.8 SUMMARY OF THE SUBROUTINE FUNCT(N,X,Q,DQ)

This subroutine evaluates Q and grad.Q, and is needed when using the Fletcher-Powell algorithm.

1) Calculate H_i $i=1,2,\dots,M$

2) Calculate

$$A^* = \frac{\sum_{i=1}^M |H_i| \cdot Y_i^d}{\sum_{i=1}^M |H_i|^2}$$

3) Calculate $E_i = A^* |H_i| - Y_i^d$ $i=1,2,\dots,M$

4) Calculate

$$Q = \sum_{i=1}^M E_i^2$$

5) Calculate

$$\frac{\partial |H_i|}{\partial \psi_n} \quad n=1,2,\dots,4K+2L$$

(Formulas are given in previous section)

6) Calculate

$$\frac{\partial Q}{\partial \psi_m} = 2 A^* \sum_{i=1}^M E_i \frac{\partial |H_i|}{\partial \psi_m} \quad m=1,2,\dots,4K+2L$$

This subroutine was written in double precision complex arithmetic in Fortran IV, and was used in conjunction with the subroutine DFMFP (double-precision Fletcher-Powell algorithm) sublied by IBM.

4.9 EXAMPLE

Design an equalizer, whose specified gain response in the frequency range $1000/2 \leq f \leq 2000/2$, is given by

$$\text{Gain} = -8 + (9/1000) \cdot \omega$$

The order of the filter is to be three ($K=1, L=1$).

$$\Psi = (f_k, \alpha_k, p_k, \theta_k; \sigma_{zj}, \sigma_{pj})^T$$

was choosen to be

$$\Psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$$

After 40 iterations

$$Q = 0.0144867$$

$$A^* = 31.4623$$

$$\begin{array}{l} \text{Poles } (-2495.37 \pm j65.5780) \\ \quad \quad (-73.3031 \quad \quad \quad) \end{array}$$

Zeros $(-42.43711 \pm j87.60335)$
 (-14.43188)

If we assume the filter to have the following form :

$$G(s) = A \cdot \frac{s^2 + a_1 s + b_1}{s^2 + c_1 s + d_1} \cdot \frac{s + \sigma_{z1}^2}{s + \sigma_{p1}^2}$$

then,

$$a_1 = 84.8723$$

$$b_1 = 76.9235 \cdot 10^4$$

$$c_1 = 49.90737 \cdot 10^2$$

$$d_1 = 6.231164 \cdot 10^6$$

$$\sigma_{z1}^2 = 14.43188$$

$$\sigma_{p1}^2 = 73.30317$$

CHAPTER V

DESIGN OF ANALOG FILTERS USING THE BILINEAR TRANSFORM

5.1 INTRODUCTION

In the design of analog filters , using optimization techniques, one encounters the problem of the infinite frequency axis. For a design specified over the complete frequency range , the frequency axis has to be divided into an infinite number of samples , specifying the magnitude at each frequency sample.

An interesting solution to this problem is to use the bilinear transformation to map the infinite frequency axis into a finite contour (circle) in the z-plane.

The problem is now reduced to one of choosing the analogue filter specifications (including the gain of the filter required at infinity) and transforming these specifications to the z-plane.

A solution is now sought for a digital filter with these transformed specifications and the solution transformed back to the s-plane.

The bilinear transform , which maps the $j\omega$ axis in the s-plane onto the unit circle in the z-plane, is given by

$$s = \frac{z - 1}{z + 1} \quad (5.1)$$

For every, $s=j\omega_a$ there is a corresponding $z=e^{j\omega_d T}$ (5.2)
 where, ω_a = analog frequency

ω_d = digital frequency

Therefore,

$$j\omega_a = \frac{e^{j\omega_d T} - 1}{e^{j\omega_d T} + 1}$$

From which

$$\omega_{a1} = \tan(\omega_{d1} T/2) = \tan(\pi \omega_1/2) \quad (5.3)$$

where ω_1 is given in fractions of the Nyquist rate.

5.2 FREQUENCY SCALING OF THE ANALOG FILTER

Suppose $\omega_1=1.0$, then

$$\omega_a = \tan(\pi/2) = \infty \text{ rad./sec.} = \infty \text{ Hz.}$$

Now, consider $\omega_1=0.9$, then

$$\omega_a = \tan(\pi \cdot 0.9/2) = \tan 81^\circ = 6.314 \text{ rad./sec.} \approx 1 \text{ Hz.}$$

That is, frequencies ranging from 1 Hz. to infinite Hz. on the $j\omega$ axis of the s -plane, are all located in a period of 0.1 the Nyquist period. The design of a digital filter with desired specifications in such a small range is impracticable.

To remedy this effect we specify the desired filter at certain samples, then linearly scale the frequency axis such that the maximum frequency sample is scaled (say) to 1 rad./sec.. Thus the filter specifications when transferred into the z -domain are all located between 0.0, and 0.5 the Nyquist rate.

Samples between zero Hz., and the minimum frequency, and between the maximum frequency and infinite Hz, are specified as desired on the digital axis.

After the filter is designed in the z-domain, the solution is transformed to the s-domain and rescaled to the desired specifications.

The method of frequency scaling is shown with the following example.

Consider a band-pass filter having the transfer function

$$Y(s) = \frac{s}{s^2 + as + b} \quad (5.4)$$

This filter has poles at

$$s = \frac{-a}{2} \pm j \sqrt{b - \frac{a^2}{4}} \quad (5.5)$$

Its center-frequency

$$\omega_c = \sqrt{b} \quad (5.6)$$

and band-width

$$BW = a \quad (5.7)$$

Now, if we define the s'-plane, where $s' = Ks$, then the transformation of the singularities in the s-plane to the

s' -plane is

$$s' = Ks = -\frac{(Ka)}{2} \pm j \sqrt{(K^2b) - \frac{Ka^2}{2}}$$

Therefore, if s is replaced by s' in Eq.(5.4), and 'a' by Ka , and 'b' by K^2b , then it is obvious from equations (5.6), and (5.7) that w_c is shifted K times, and that the band-width is multiplied by K .

5.3 PROCEDURE

- 1) Define the required analog filter on the frequency axis at a finite number of samples.
- 2) Divide the prescribed frequencies by a scale factor, such that the maximum frequency specified is 1 rad./sec.
- 3) Using eq.(5.3),

$$w_i = \frac{2}{\pi} \tan^{-1}(w_{ai})$$

and the bilinear transformation is introduced to transform the analog frequency axis to the digital frequency axis.

- 4) Frequencies between the maximum frequency specified and infinite Hz are transferred between 0.5 and 1.0 the Nyquist rate on the digital frequency axis. This range is now divided into a finite number of frequencies, specifying at each frequency the desired amplitude at the higher ranges of frequencies in the s -domain.

5) Design the digital filter to meet the required magnitude response using the outlined optimization procedure.

6) Replace z by $\frac{1+s}{1-s}$ in the digital filter designed.

If the digital filter designed had the form

$$Y(z) = A \prod_{k=1}^K \frac{z^2 + a_k z + b_k}{z^2 + c_k z + d_k} \prod_{j=1}^L \frac{z - a_j}{z - b_j}$$

Then,

$$Y(s) = A' \prod_{k=1}^K \frac{s^2 + a'_k s + b'_k}{s^2 + c'_k s + d'_k} \prod_{j=1}^L \frac{s + a'_j}{s + b'_j}$$

where ,

$$A' = A \prod_{k=1}^K \frac{1 - a_k + b_k}{1 - c_k + d_k} \prod_{j=1}^L \frac{1 + a_j}{1 + b_j}$$

$$= \frac{\sum_{i=1}^M |H(s_i, \Psi)| \cdot Y_i^d}{\sum_{i=1}^M |H(s_i, \Psi)|^2} \quad (\text{Chapter IV})$$

$$a'_k = \frac{2 - 2b_k}{1 - a_k + b_k}$$

$$b'_k = \frac{1 + a_k + b_k}{1 - a_k + b_k}$$

$$c'_k = \frac{2 - 2d_k}{1 - c_k + d_k}$$

$$d'_k = \frac{1 + c_k + d_k}{1 - c_k + d_k}$$

$$a'_j = \frac{a_j - 1}{a_j + 1}$$

$$b'_j = \frac{b_j - 1}{b_j + 1}$$

7) If the frequency division scale factor is λ ,
then the desired analog filter should have the parameters

$$a'_k = \lambda a'_k$$

$$b'_k = \lambda^2 b'_k$$

$$c'_k = \lambda c'_k$$

$$d'_k = \lambda^2 d'_k$$

$$a'_j = \lambda a'_j$$

$$b'_j = \lambda b'_j$$

5.4 EXAMPLES

Example 1: Design a high-pass filter with maximum magnitude 1.0, and a value of 0.5 at 10 KHz.

In this example we shall alter the previous procedure presented, to show the method of scaling.

1- We first design a recursive digital high-pass filter having cut-off at 0.5 the Nyquist period.

The filter specifications are chosen to be

$$w = 0.00, 0.495(0.05); \quad Y^d = 0.0$$

$$w = 0.5; \quad Y^d = 0.5$$

$$w = 0.505, 1.0(0.05); \quad Y^d = 1.0$$

Ψ is chosen to be $(0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$.

For $K=L=1$, and after 10 iterations the results are

$$Q=0.5968$$

$$A^* = 0.30021$$

$$a_1 = -1.23881$$

$$b_1 = 0.998895$$

$$c_1 = -0.30137$$

$$d_1 = 0.0406117$$

$$a_{j1} = 0.89988$$

$$b_{j1} = 0.24049$$

Poles $(0.15068 \pm j0.13381)$
 $(0.24049 \quad)$

Zeros $(0.61940 \pm j0.78436)$
 $(0.89988 \quad)$

2- The filter is then transferred to the s-plane , and we get the following results

$$A = 1.12269$$

$$a_1 = 0.68598 \times 10^{-3}$$

$$b_1 = 0.23476$$

$$c_1 = 1.42981$$

$$d_1 = 0.55086$$

$$a_{j1} = 0.5269 \times 10^{-1}$$

$$b_{j1} = 0.61226$$

Poles $(-0.71490 \pm j0.48452)$
 $(-b_{j1} \quad)$

Zeros $(-0.34299 \times 10^{-3} \pm j0.199425)$
 $(-a_{j1} \quad)$

3- The scale factor is obtained as follows

$$w_a = \tan(w\pi/2) = \tan(\pi/4) = 1.0$$

$$f_a = 1.0/2\pi$$

$$\lambda = 10^4 \times 2\pi$$

4- The desired filter parameters are

$$a'_1 = \lambda a_1$$

$$b'_1 = \lambda^2 b_1$$

$$c'_1 = \lambda c_1$$

$$d'_1 = \lambda^2 d_1$$

$$a'_{j1} = \lambda a_{j1}$$

$$b'_{j1} = \lambda b_{j1}$$

The poles and zeros are each multiplied by λ .

Example 2 : Design a low-pass filter with cut-off at 10 Krad/sec.

According to the procedure given in sec.5.3

1- The filter was specified in the analog frequency domain as follows

$$w_a = 0.0, 9500 (950); \quad y^d = 1.0$$

$$w_a = 10^4; \quad y^d = 0.707$$

$$w_a = 10^4, 15000 (990); \quad y^d = 0.0$$

2- The scale factor $\lambda = 1.0/15,000$

3- The filter specifications from 0.0 to 0.5, on the digital frequency axis are obtained as follows

$$\omega_1 = \frac{2}{\pi} \tan^{-1}(\omega_{al} \cdot \lambda)$$

4- The filter specifications from 15000 rad/sec. (0.5 the Nyquist rate on the digital frequency axis), and infinite rad/sec (the Nyquist rate), are specified as follows

$$\omega_1 = 0.5, 1.0(0.05); \quad \gamma^d = 0.0$$

5- After designing the digital filter, starting with

$$\psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T;$$

and taking $K=L=1$, we transform the digital filter to the s-domain, and scale the frequency axis as in sec. 5.3 we obtain the required analog filter.

After 10 iterations the analog filter parameters were

$$Q = 1.07466$$

$$A^* = 0.052539$$

$$a_1 = 0.318772 \times 10^5$$

$$b_1 = 0.254039 \times 10^9$$

$$c_1 = 0.97429 \times 10^4$$

$$d_1 = 0.67080 \times 10^8$$

$$a_{j1} = 0.36211 \times 10^5$$

$$b_{j1} = 0.73302 \times 10^4$$

Poles $(-0.48714 \times 10^4 \pm j0.62515)$
 $(-b_{j1})$

Zeros $(-0.159386 \times 10^5 \pm j0.65840 \times 10^4)$
 $(-a_{j1})$

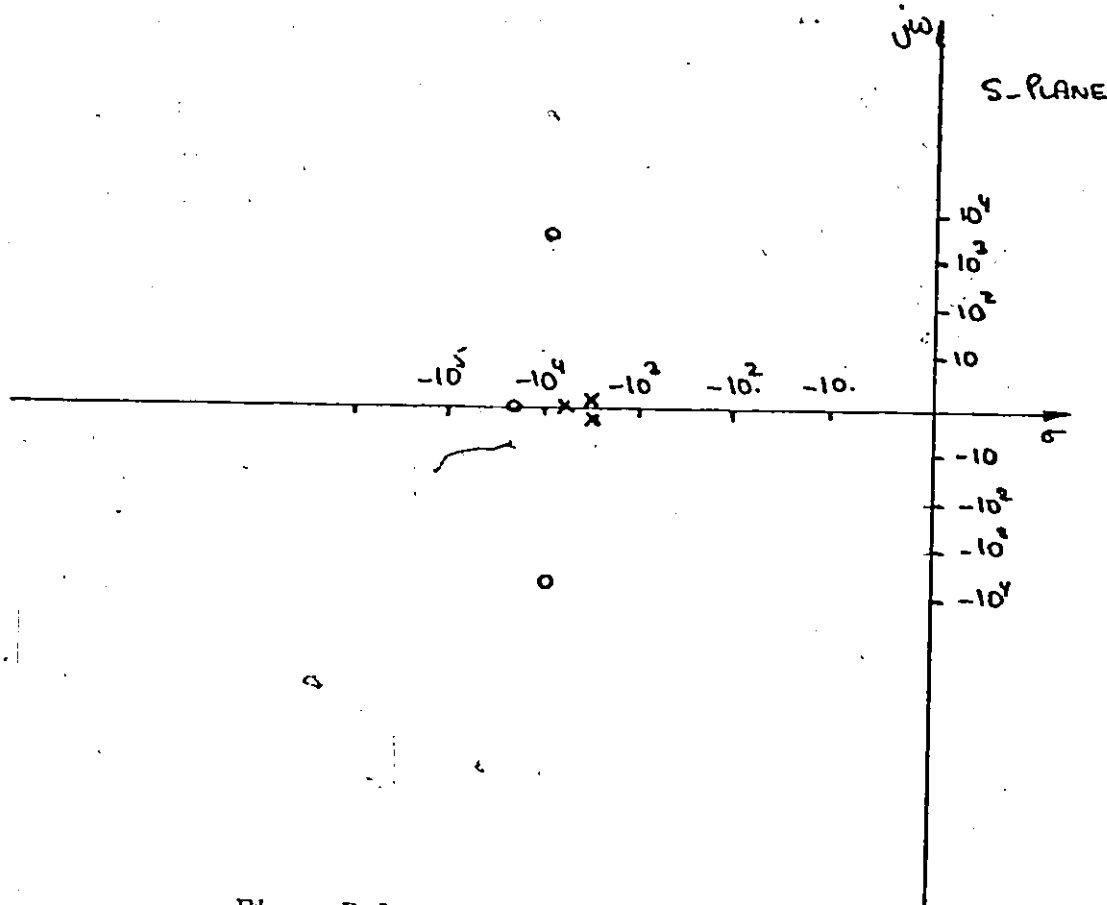


Fig. Pole-zero location of the third order low-pass filter

After 30 iterations

$$Q = 0.49738$$

$$A = 0.027621$$

$$a_1 = 0.31860 \times 10^5$$

$$b_1 = 0.25376 \times 10^9$$

$$c_1 = 0.37639 \times 10^4$$

$$d_1 = 0.58154 \times 10^8$$

$$a_{j1} = 0.25196 \times 10^5$$

$$b_{j1} = 0.26698 \times 10^4$$

$$\text{Poles } (-0.188197 \times 10^4 \pm j0.456155 \times 10^4) \\ (-b_{j1})$$

$$\text{Zeros } (-0.1593000 \times 10^5 \pm j0.739004 \times 10^4) \\ (-a_{j1})$$

For an eighth-order filter $K=3$, $L=2$ was chosen, and ψ_0 was chosen from the final conditions of the previous design.

$\psi_0 = (0.687 \times 10^{-2}, 0.428, 0.876, 0.843, \text{repeated } 3 \text{ times since } K=3); -0.399, 0.379, \text{ (repeated } 2 \text{ times, since } L=2 \text{)}).$

After 17 iterations the results of the analog filter were

$$Q=0.22631$$

$$A=0.19915 \times 10^{-3}$$

$$a_1=0.3187 \times 10^5$$

$$a_2=a_3=a_1$$

$$b_1=0.25408 \times 10^9$$

$$b_2=b_3=b_1$$

$$c_1=0.70117 \times 10^4$$

$$c_2=c_3=c_1$$

$$d_1=0.78435 \times 10^8$$

$$d_2=d_3=d_1$$

$$\text{Poles 1 } (-0.35059 \times 10^4 \pm j0.120843 \times 10^{-2})$$

=complex poles of one cascade

=Poles 2 =Poles 3

Zeros 1 $(-0.159399 \times 10^5 \pm j0.813286 \times 10^4)$

=complex zeros of one cascade

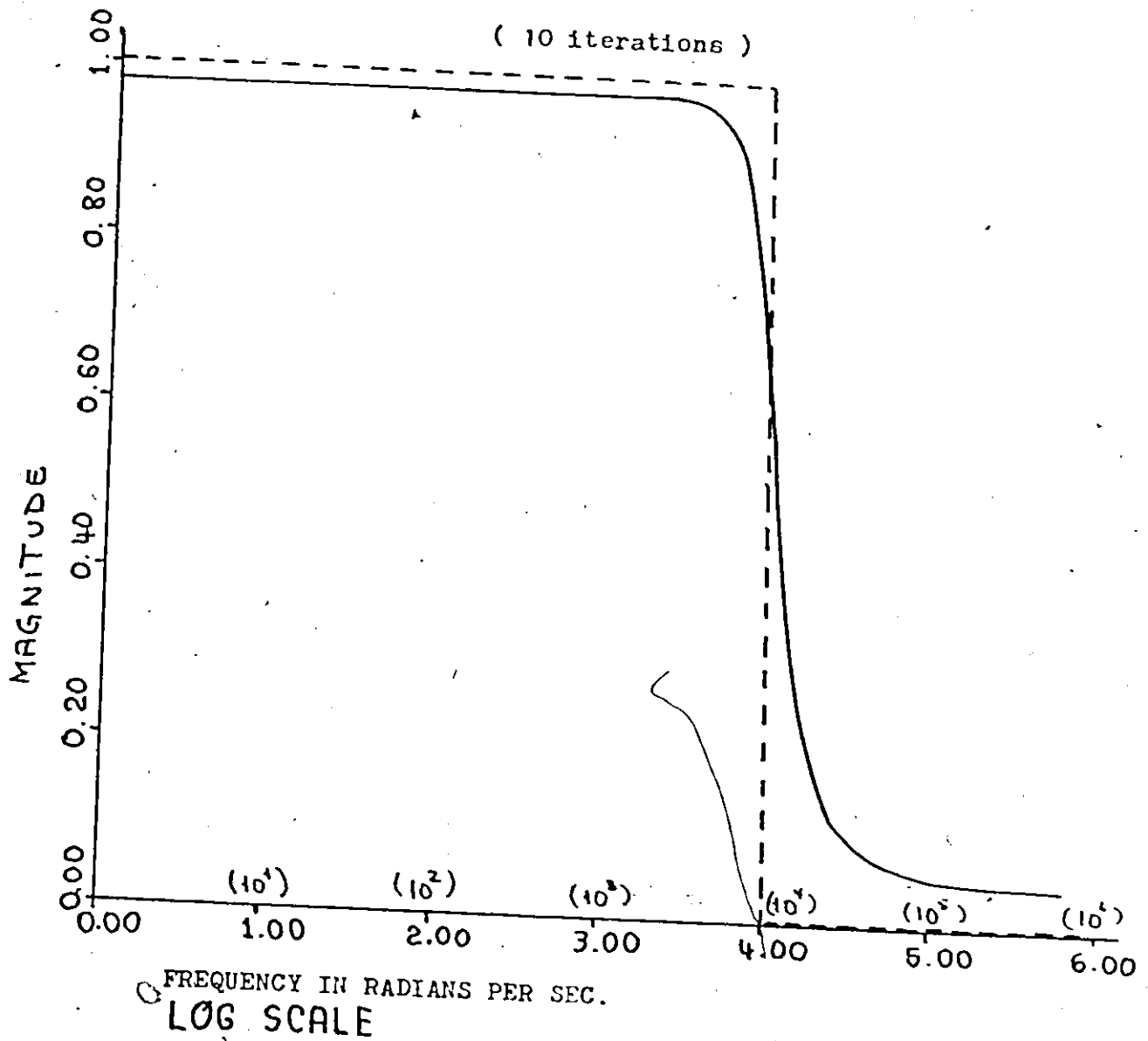
=Zeros 2 =Zeros 3

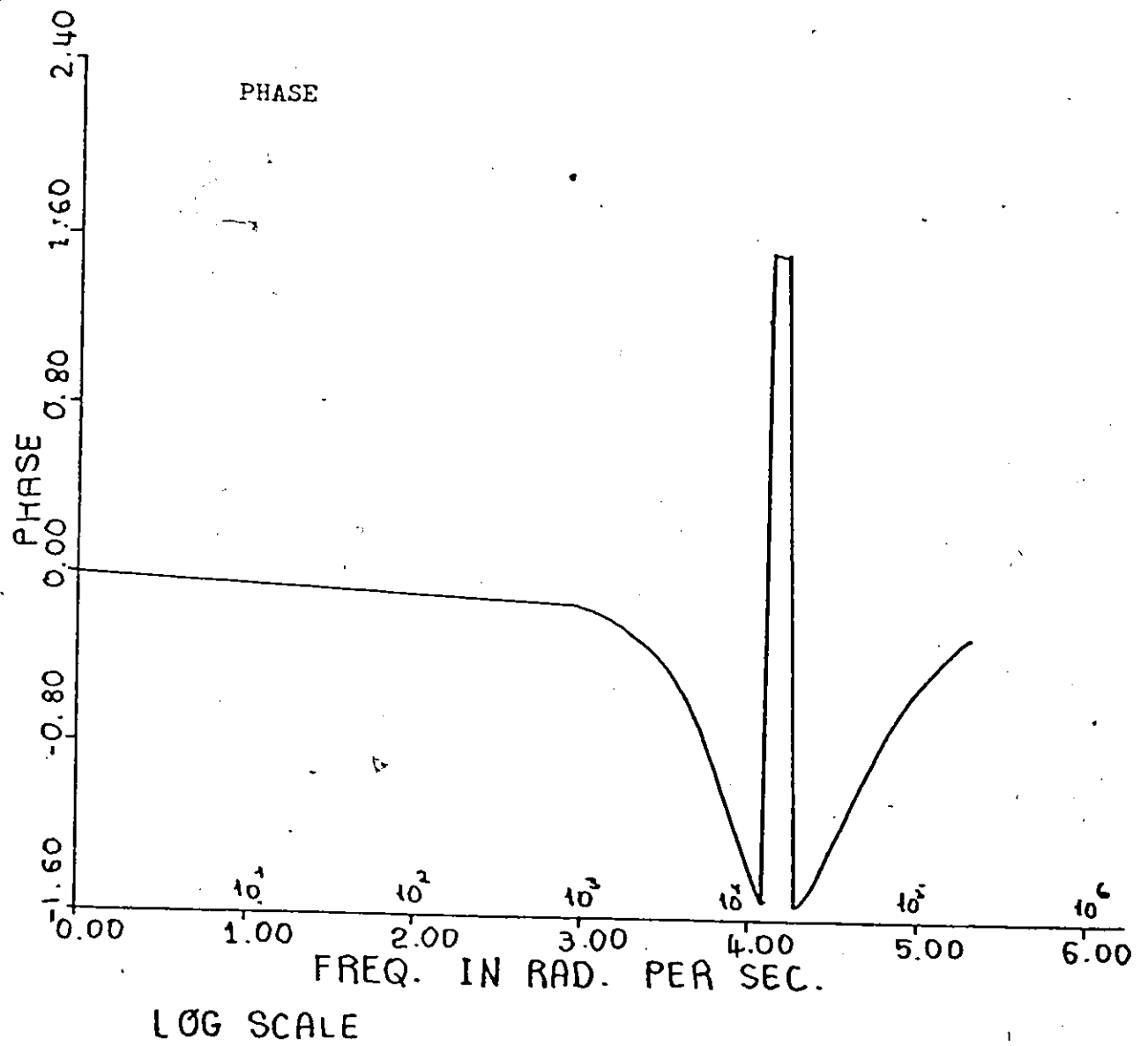
$$a_{j1} = 0.492725 \times 10^5 = a_{j2}$$

$$b_{j1} = 0.402290 \times 10^4 = b_{j2}$$

Example 2 : Third order low-pass filter

$$Q = 1.07466$$

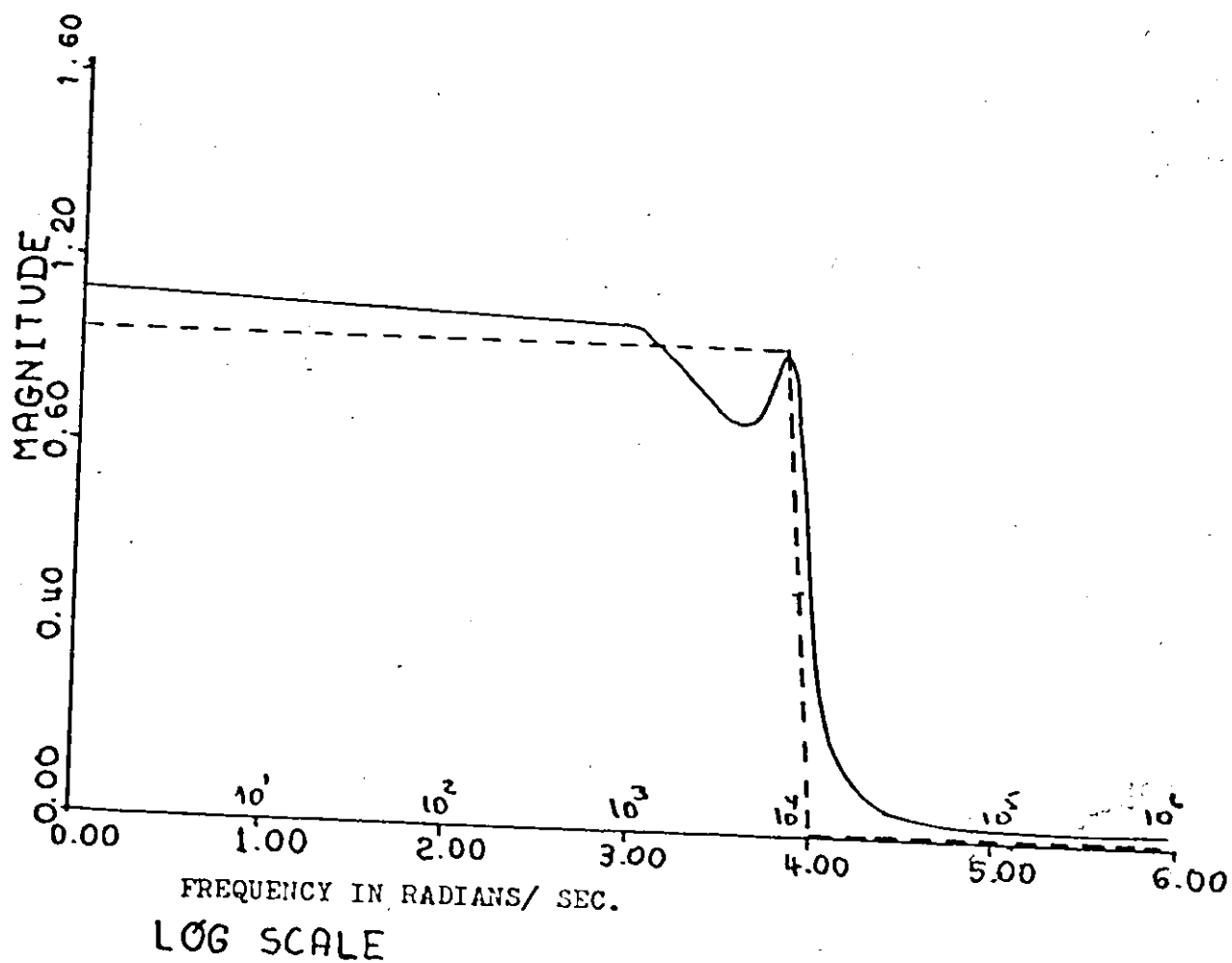


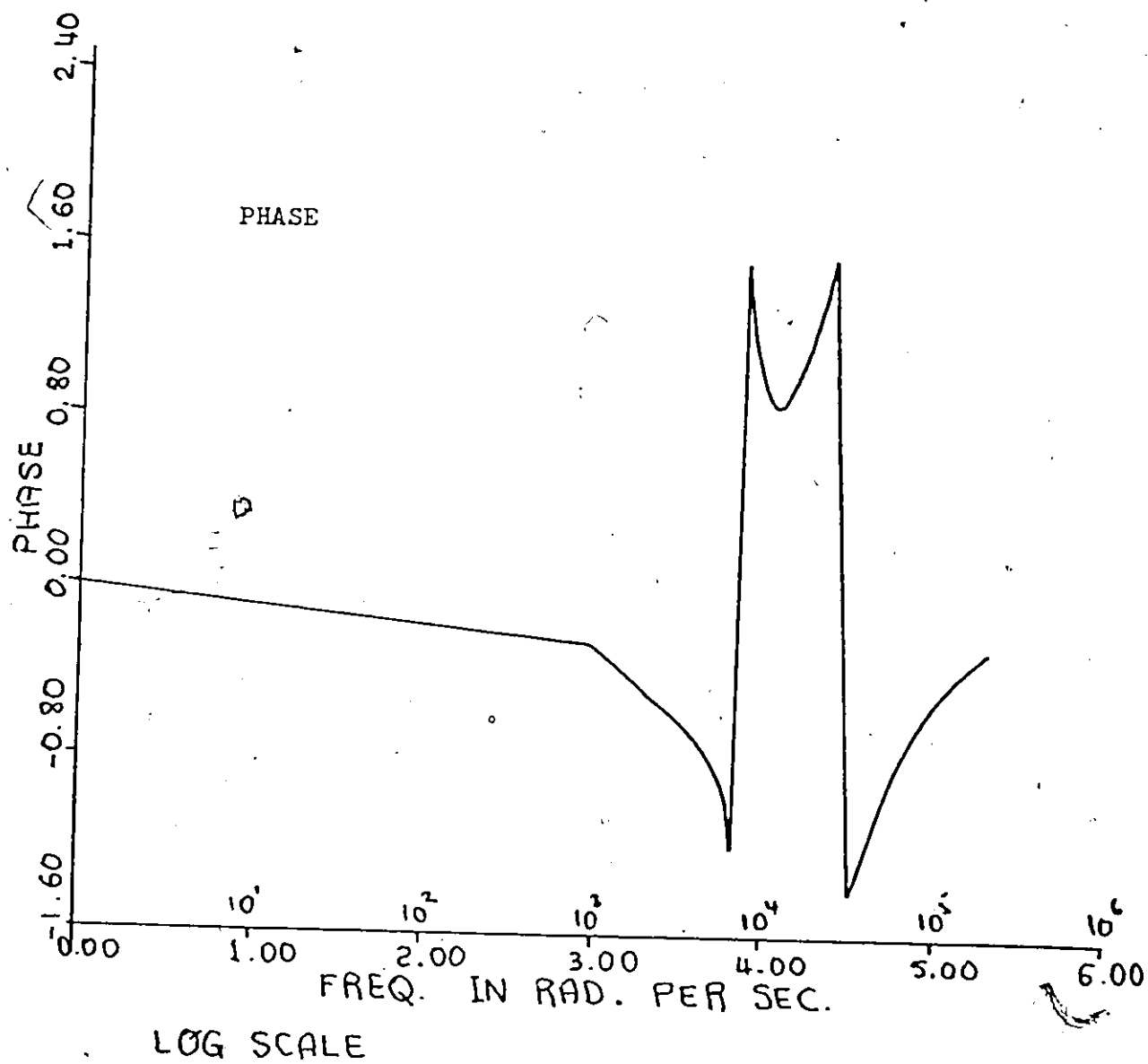


Example 2 : Third order low-pass filter

$$Q = 0.49738$$

(30 iterations)

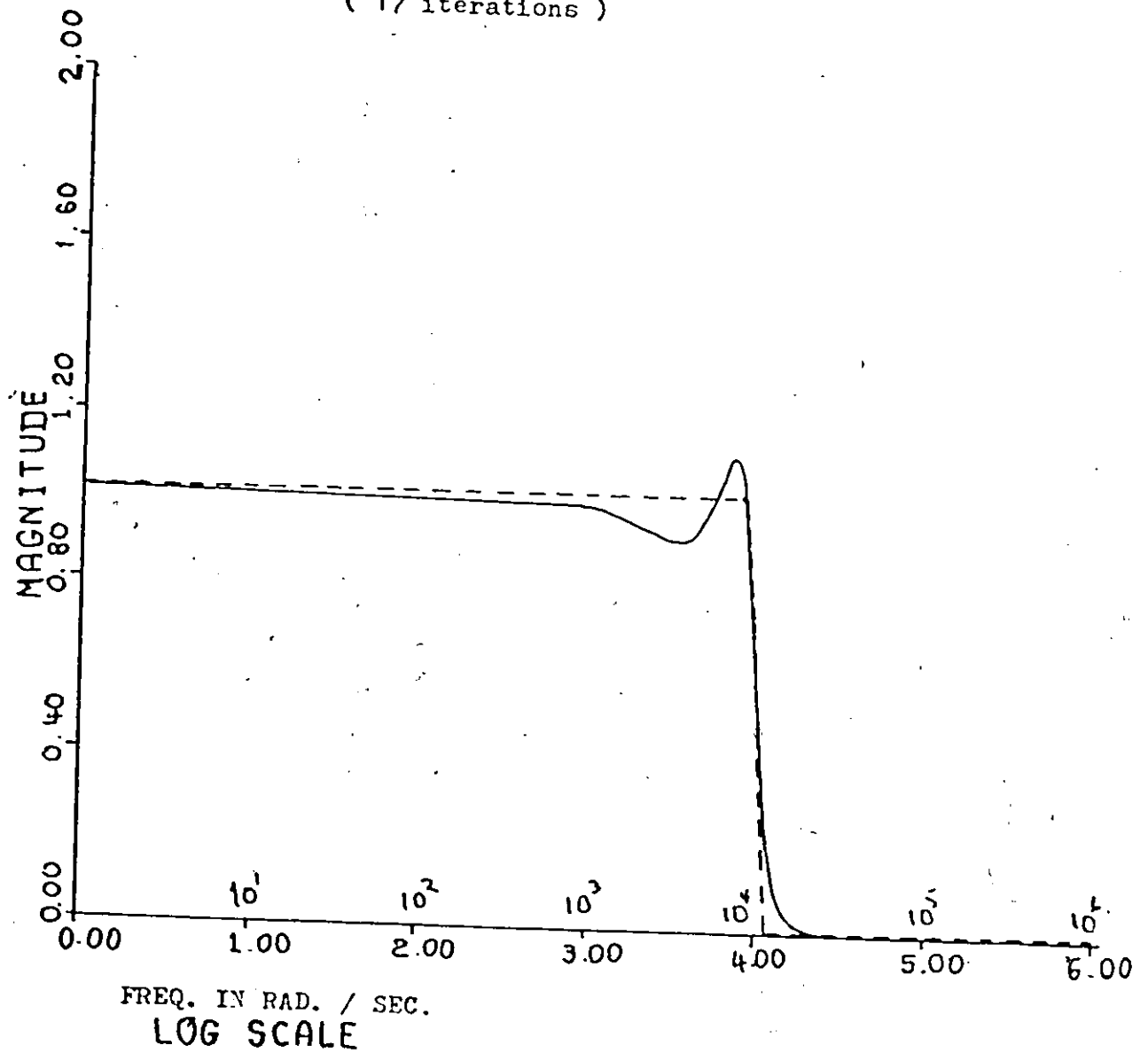


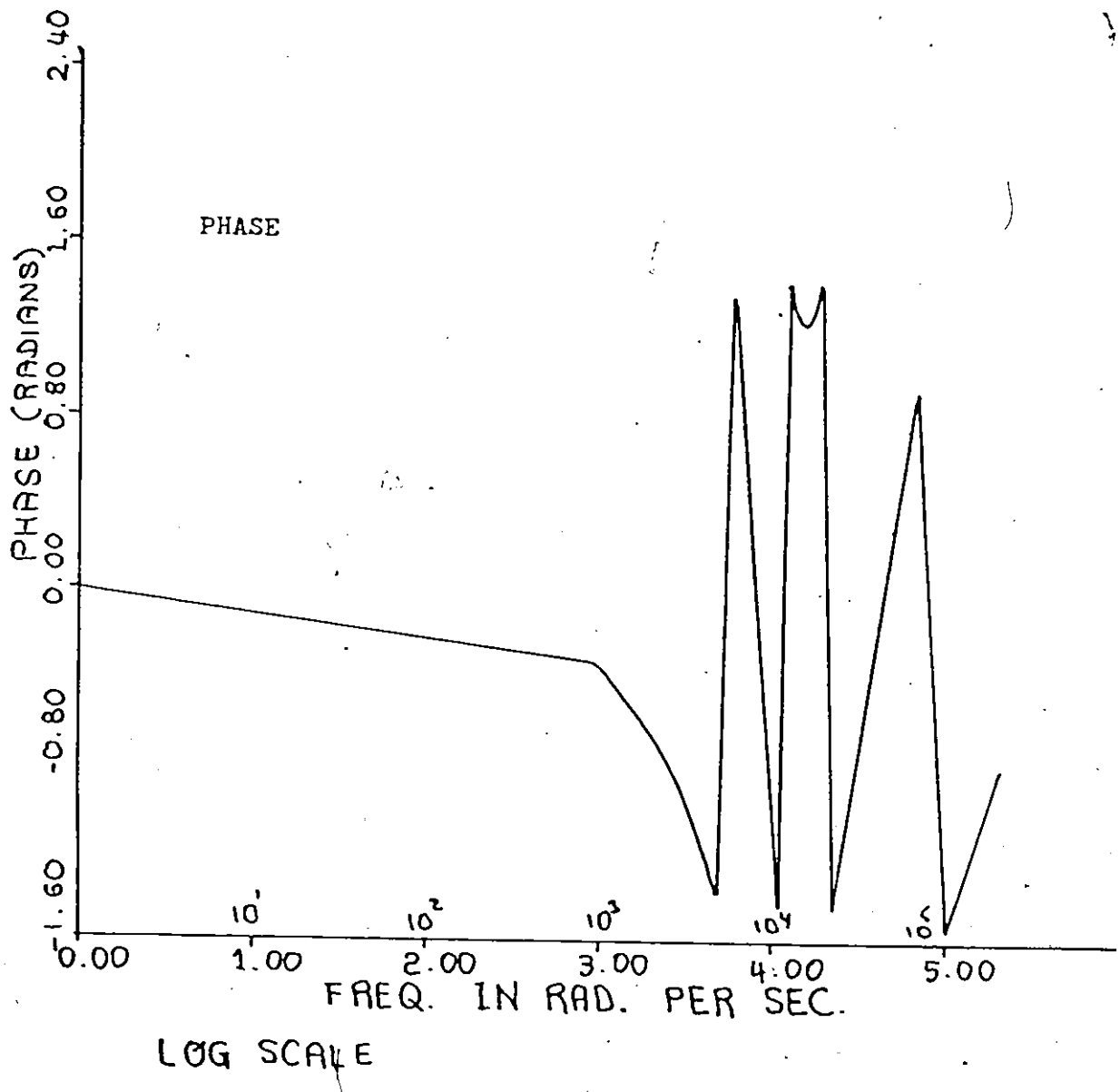


Example 2 : Eighth order low-pass filter

$$Q = 0.22632$$

(17 iterations)





Example 3 : Design a high-pass filter having the following specifications

$$\begin{aligned} \omega_a &= 0.0, 9000(1000); & Y^d &= 0.0 \\ \omega_a &= 10^4; & Y^d &= 0.707 \\ \omega_a &= 13000, 30000(3000); & Y^d &= 1.0 \end{aligned}$$

- 1- λ = scale factor = $1.0/30000$
- 2- $\omega_1 = \frac{2}{\pi} \tan^{-1}(\omega_{ai} \cdot \lambda)$
- 3- Frequencies between 0.5 and 1.0 times the Nyquist rate, which correspond to 30000 rad/sec., and infinite rad./sec. are selected as follows

$$\omega_1 = 0.505, 1.0(0.05); \quad Y^d = 1.0$$

- 4- Proceeding as in example 2, Ψ_0 is chosen to be

$$\Psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$$

For $K=L=1$, and after 60 iterations, the results for the analog filter desired are

$$\begin{aligned} Q &= 1.0060 \\ A &= 0.8941 \\ a_1 &= 0.6795 \times 10^5 \\ b_1 &= 0.1166 \times 10^{10} \\ c_1 &= 0.4069 \times 10^5 \\ d_1 &= 0.8354 \times 10^9 \end{aligned}$$

$$a_{j1} = 0.49831 \times 10^2$$

$$b_{j1} = 0.29753 \times 10^5$$

$$\text{Poles } (-0.20348 \times 10^5 \pm j0.34195 \times 10^4) \\ (-b_{j1})$$

$$\text{Zeros } (-0.33979 \times 10^5 \pm j0.20529 \times 10^5) \\ (-a_{j1})$$

For the same filter specifications, except that at

$$w_a = 10^4; \quad Y^d = 0.50$$

and $K=3, L=2$ is chosen.

ψ_0 was chosen as the final value obtained from the previous case

$\psi_0 = (-0.31074, 0.56274, 0.73113, 0.01135, (\text{repeated } 3 \text{ times, since } K=3); 1.50023, 0.14744, (\text{repeated } 2 \text{ times, since } L=2))$.

After 30 iterations the results obtained are.

$$Q = 0.43007$$

$$A = 0.92021$$

$$a_1 = 0.623451 \times 10^5$$

$$b_1 = 0.99282 \times 10^9$$

$$c_1 = 0.15352 \times 10^5$$

$$d_1 = 0.25867 \times 10^9$$

$$a_2 = 0.61965 \times 10^5$$

$$b_2 = 0.97905 \times 10^9$$

$$c_2 = 0.65973 \times 10^5$$

$$d_2 = 0.11172 \times 10^{10}$$

$$a_3 = 0.62762 \times 10^5$$

$$b_3 = 0.10077 \times 10^{10}$$

$$c_3 = 0.72765 \times 10^5$$

$$d_3 = 0.13466 \times 10^{10}$$

$$a_{j1} = 0.3982 \times 10^3$$

$$b_{j1} = 0.2494 \times 10^5$$

$$a_{j2} = 0.3982 \times 10^3$$

$$b_{j2} = 0.2366 \times 10^5$$

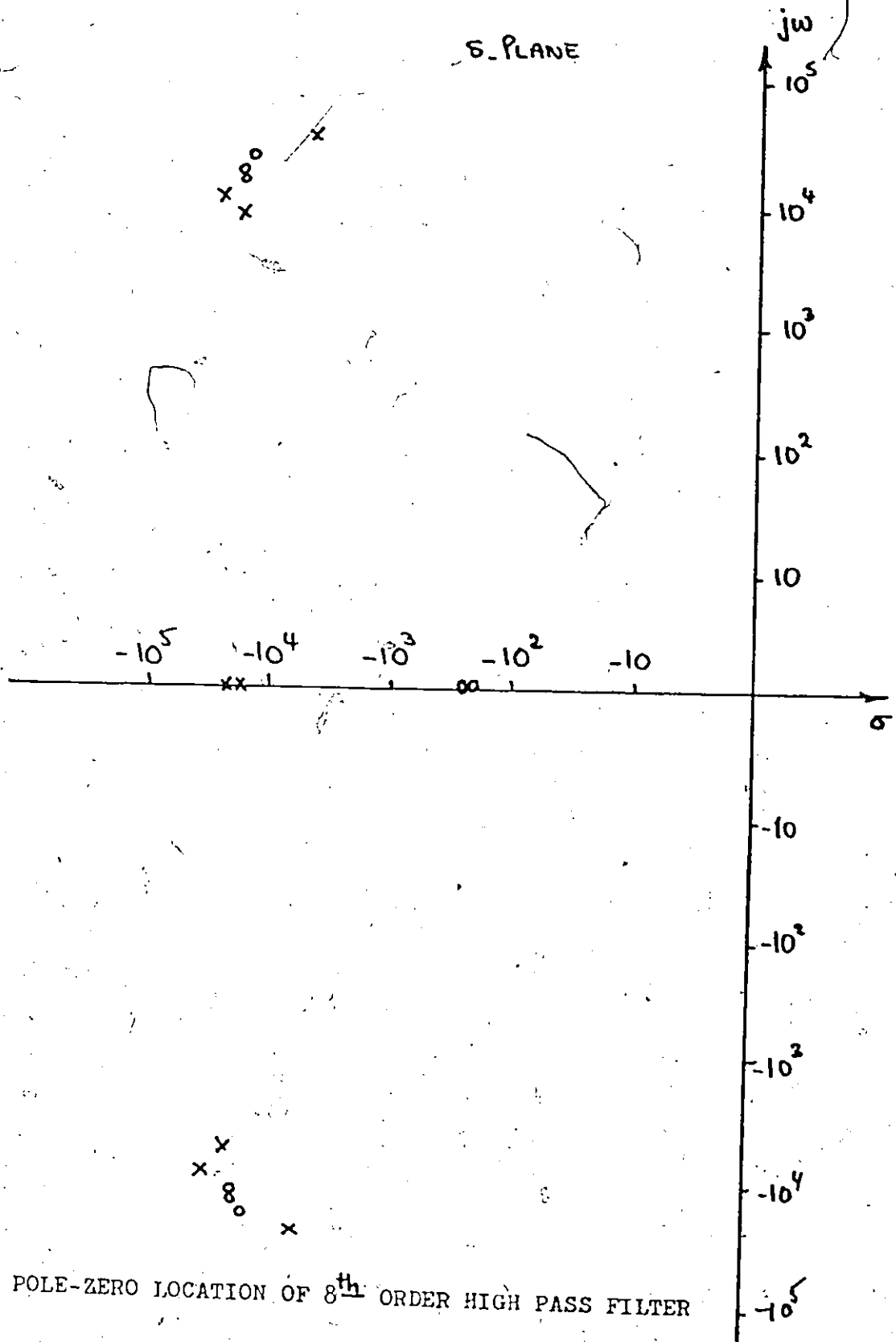
Poles

$$\begin{aligned} & (-0.76758 \times 10^4 \pm j0.45925 \times 10^4) \\ & (-0.32987 \times 10^5 \pm j0.43749 \times 10^4) \\ & (-0.36382 \times 10^5 \pm j0.47948 \times 10^4) \\ & (-b_{j1}, \text{ and } -b_{j2}) \end{aligned}$$

Zeros

$$\begin{aligned} & (-0.311725 \times 10^5 \pm j0.1413 \times 10^5) \\ & (-0.309824 \times 10^5 \pm j0.5393 \times 10^4) \\ & (-0.313809 \times 10^5 \pm j0.4789 \times 10^4) \\ & (-a_{j1}, \text{ and } -a_{j2}) \end{aligned}$$

S. PLANE

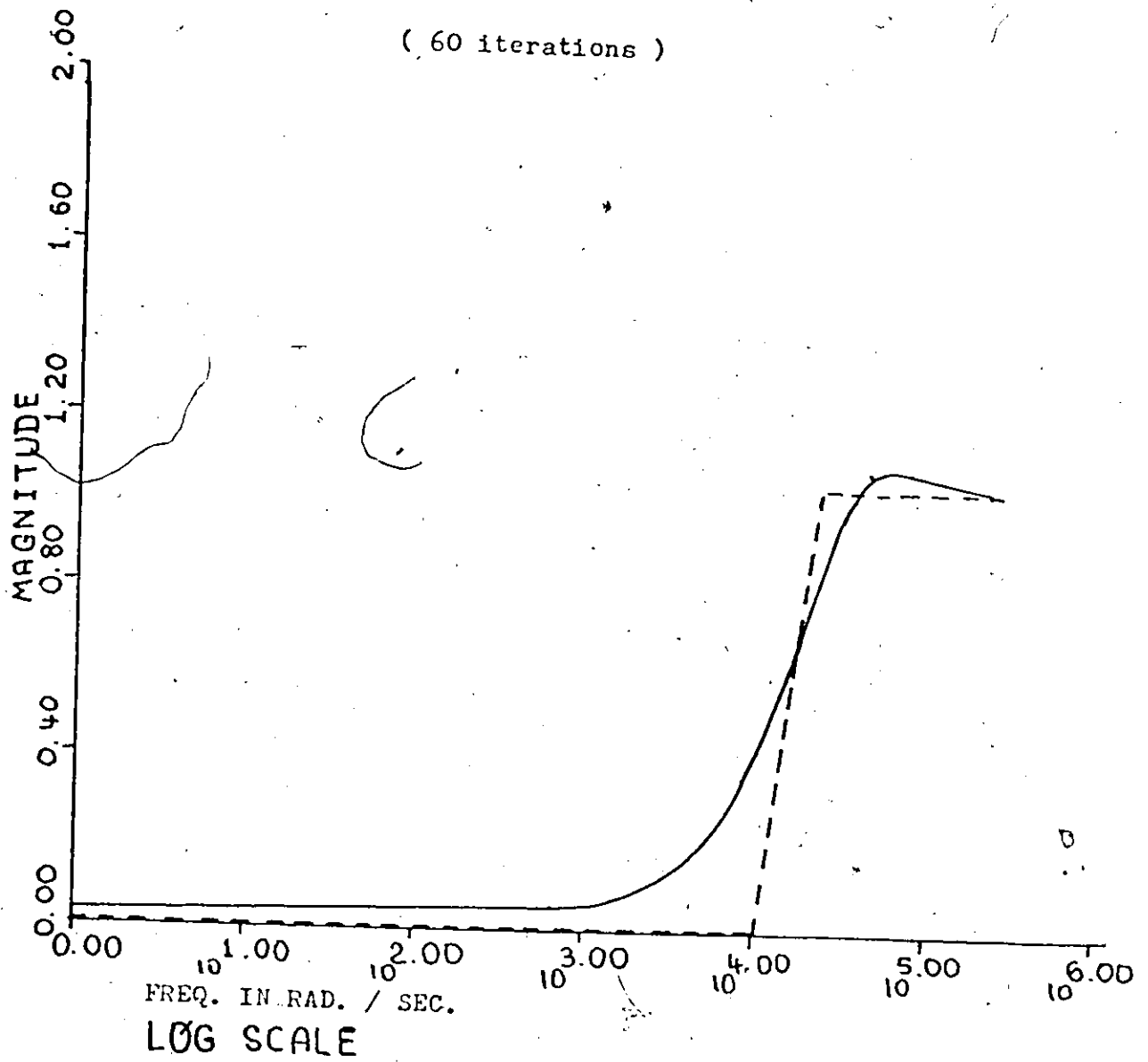


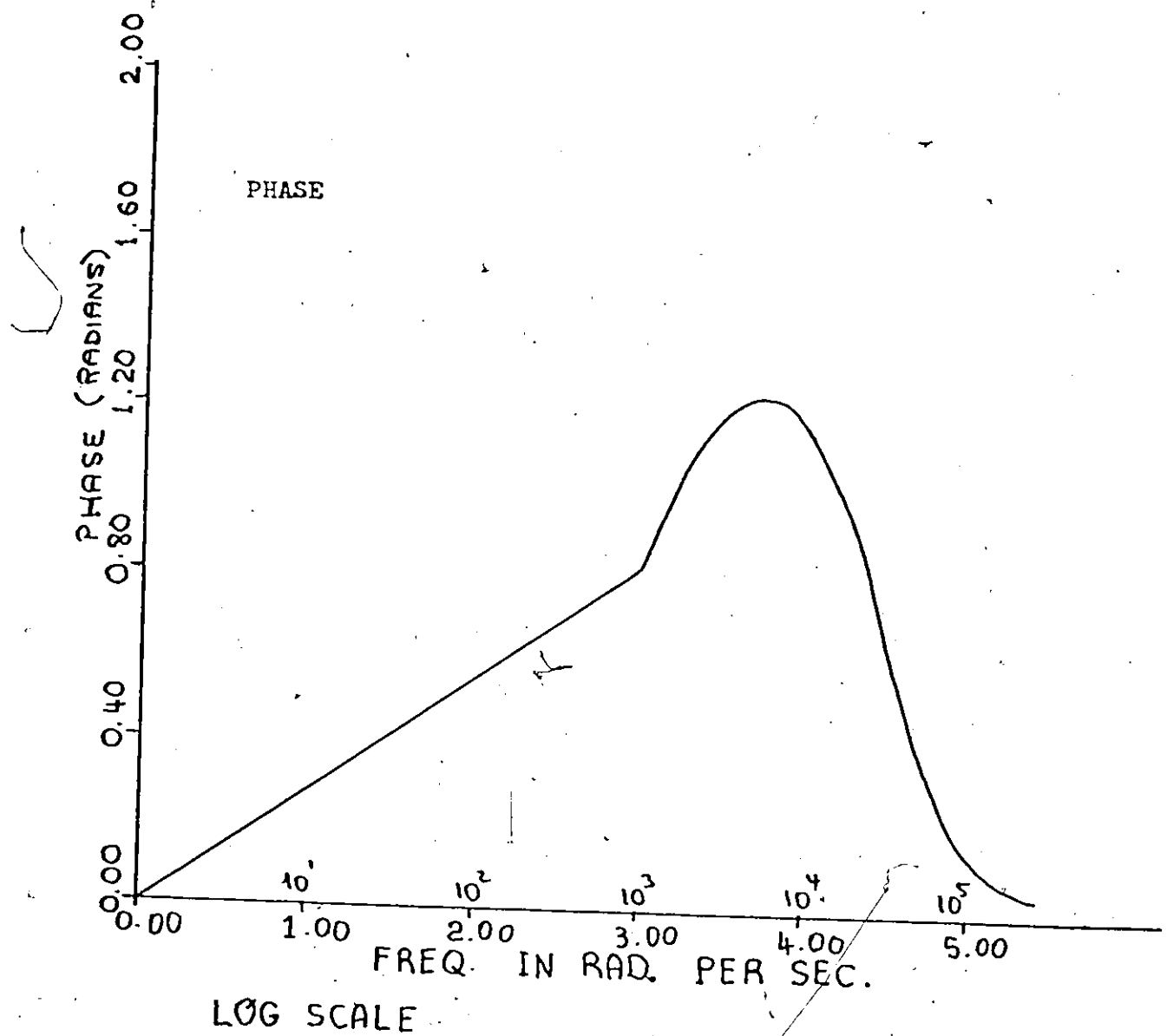
POLE-ZERO LOCATION OF 8th ORDER HIGH PASS FILTER

Example 3 : Third order high-pass filter

$$Q=1.0060$$

(60 iterations)

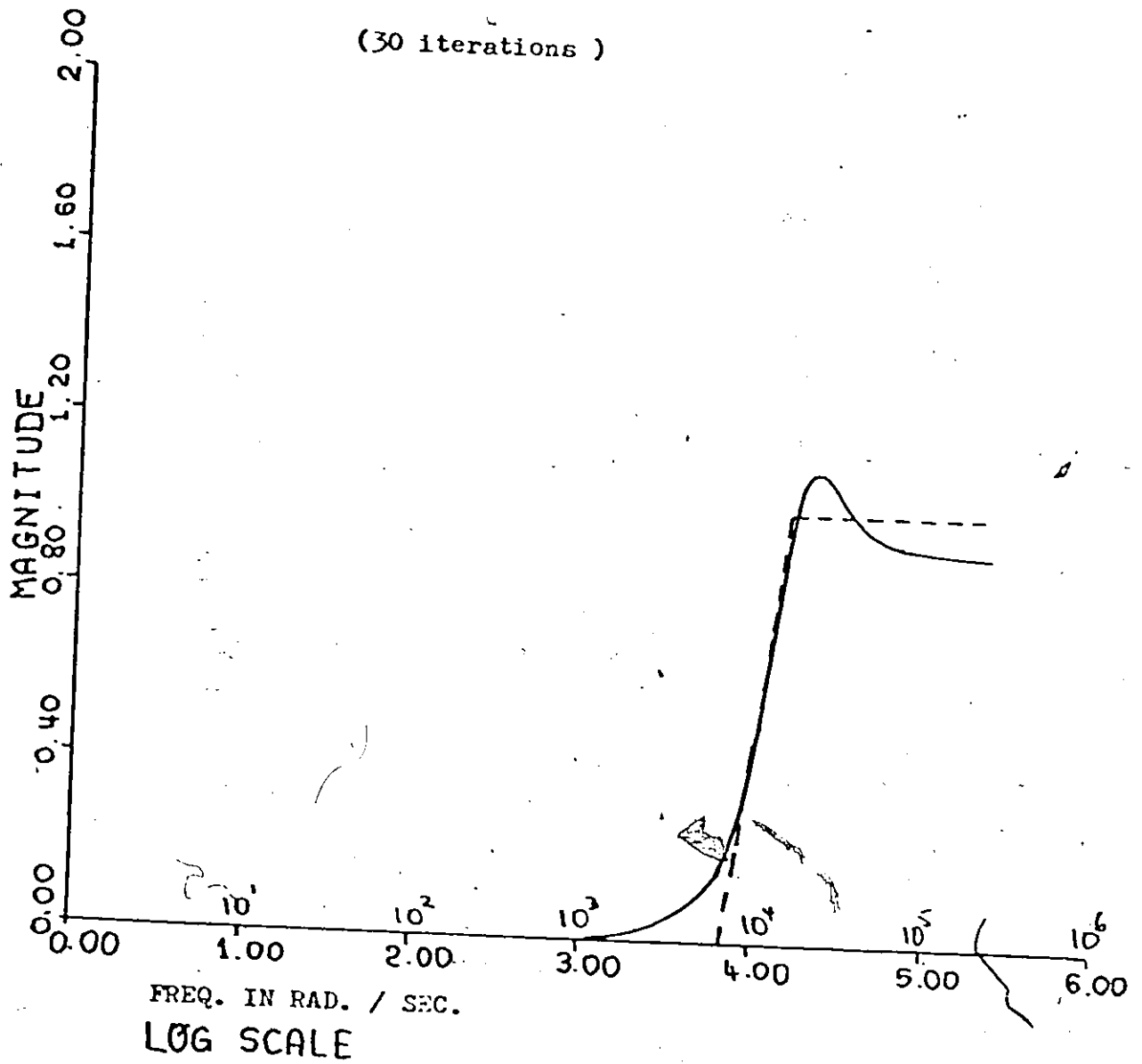


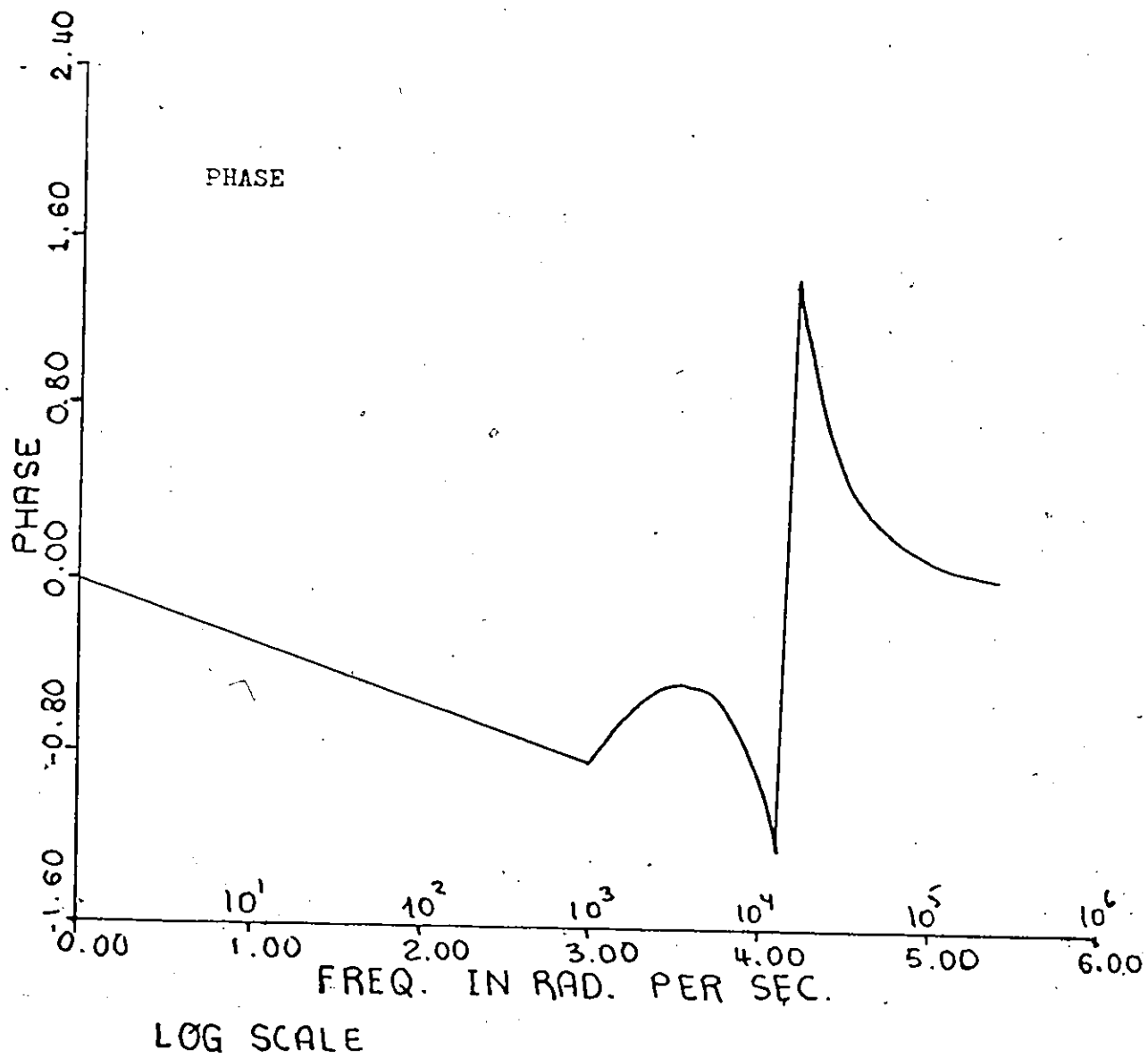


Example 3 : Eighth order high-pass filter

$$Q = 0.43007$$

(30 iterations)





CHAPTER VI

DESIGN OF DIGITAL FILTERS BY IMPLEMENTING CONSTRAINTS ON AN ANALOG FILTER

6.1 THE METHOD

Consider an analog filter having the transfer function

$$Y(s) = A \prod_{k=1}^K \frac{s^2 + a_k^2 s + b_k^2}{s^2 + c_k^2 s + d_k^2} \quad (6.1)$$

The coefficients are squared to ensure that they always remain positive, and hence the poles and zeros are confined to the left-half of the s-plane.

If the left-half of the s-plane is mapped onto the unit circle in the z-plane, and the poles and zeros of the transfer-function given by eq.(6.1) are mapped into the z-plane, then the mapped function will have its poles and zeros confined inside the unit circle.

The bilinear transformation which maps the $j\omega$ -axis, and the left-half of the s-plane onto a unit circle in the z-plane is given by

$$s = \frac{z - 1}{z + 1} \quad (6.2)$$

Hence, the transfer function of the recursive digital filter that has its poles and zeros confined inside the unit circle is obtained by substituting eq.(6.2) in eq.(6.1).

Therefore ;

$$Y(z,X) = A \prod_{k=1}^K \frac{z^2(1+a_k^2+b_k^2) + z(-2+2b_k^2) + (1-a_k^2+b_k^2)}{z^2(1+c_k^2+d_k^2) + z(-2+2d_k^2) + (1-c_k^2+d_k^2)} \quad (6.3)$$

$$= A.H(z,\psi)$$

where, $X=(a_k, b_k, c_k, d_k; A)^T$

and $\psi=(a_k, b_k, c_k, d_k)^T$

6.2 Summary of the subroutine FUNCT(N,X,Q,DQ)

1- Calculate H_i $i=1,2,\dots,M$

2- Calculate

$$A^* = \frac{\sum_{i=1}^M |H_i| Y_i^d}{\sum_{i=1}^M |H_i|^2}$$

3- Calculate $E_i = A^* |H_i| - Y_i^d$ $i=1,2,\dots,M$

4- Calculate $Q = \sum_{i=1}^M E_i^2$

5- Calculate $\frac{\partial |H_1|}{\partial \psi_n}$ $n=4K$

$$\frac{\partial |H_1|}{\partial a_k} = |H_1| \cdot \operatorname{Re} \left(\frac{2a_k z^2 - 2a_k}{N} \right)$$

$$\frac{\partial |H_1|}{\partial b_k} = |H_1| \cdot \operatorname{Re} \left(\frac{2b_k z^2 + 4b_k z + 2b_k}{N} \right)$$

$$\frac{\partial |H_1|}{\partial c_k} = |H_1| \cdot \operatorname{Re} \left(\frac{2c_k z^2 - 2c_k}{D} \right)$$

$$\frac{\partial |H_1|}{\partial d_k} = |H_1| \cdot \operatorname{Re} \left(\frac{2d_k z^2 + 4d_k z + 2d_k}{D} \right)$$

where, $N = z^2(1+a_k^2+b_k^2) + z(-2+2b_k^2) + (1-a_k^2+b_k^2)$

$D = z^2(1+c_k^2+d_k^2) + z(-2+2d_k^2) + (1-c_k^2+d_k^2)$

6.3 EXAMPLE

Consider the specifications of the ideal low-pass filter given in sec.3.9.

In this case a second order filter is considered, and the parameter vector is initially chosen as

$$\psi_0 = (0.1, 0.1, 0.1, 0.1)^T$$

After 100 iterations

$$Q = 0.53475$$

$$A^* = 0.1368$$

$$\text{Poles } (0.8957 \pm j0.19554)$$

$$\text{Zeros } (0.84124 \pm j0.54045)$$

If $K=2$ is now chosen, and the final value of ψ is taken as the initial value in this case. After 100 iterations

$$Q = 0.032563$$

$$A^* = 0.031198$$

$$\text{Poles } (0.86281 \pm j0.13583)$$

$$(0.93122 \pm j0.27786)$$

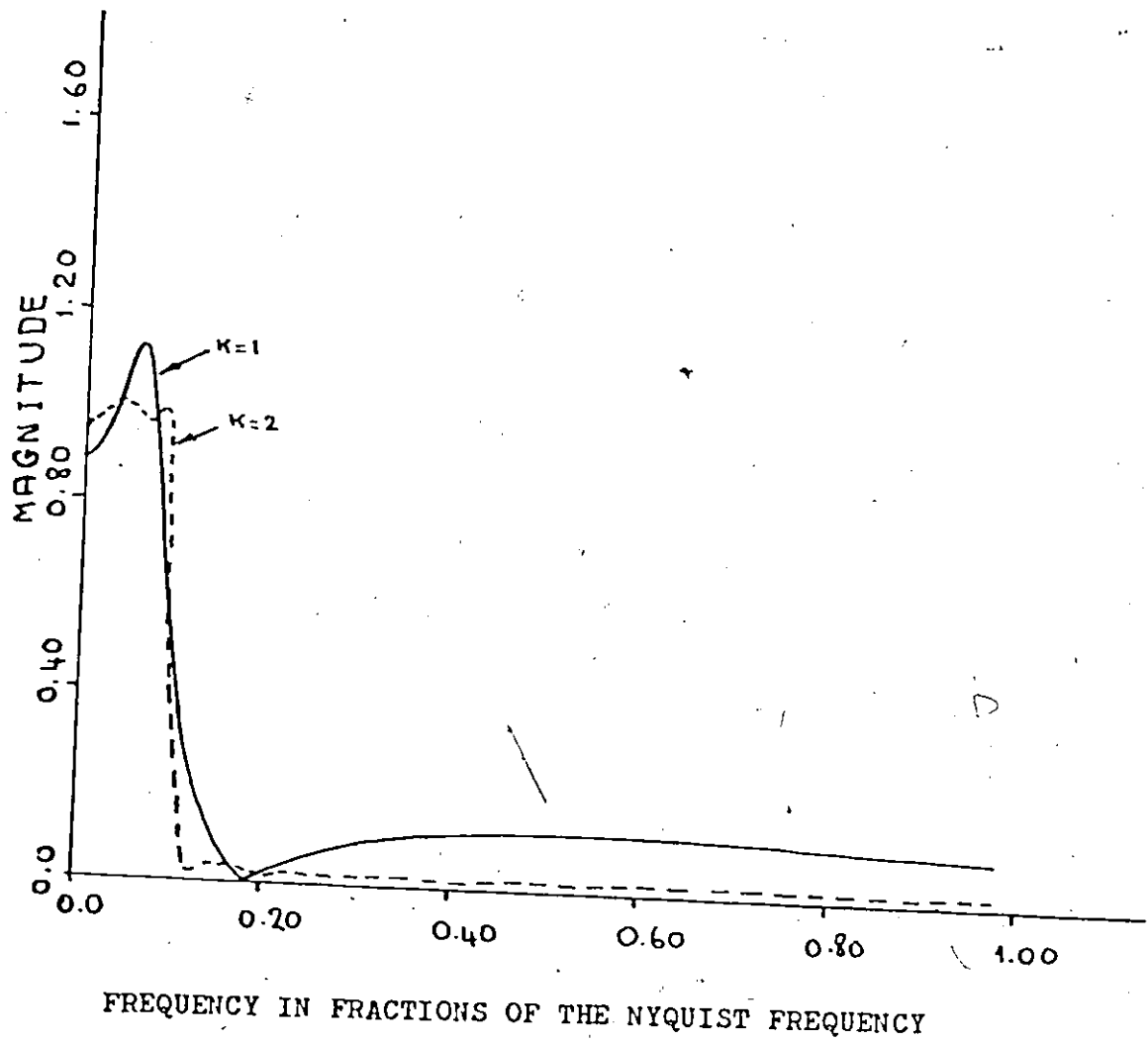
$$\text{Zeros } (0.92596 \pm j0.37744)$$

$$(0.67643 \pm j0.73636)$$

6.4 DISCUSSION

The method discussed in this chapter has two advantages over the method previously discussed in chapter III:

- 1-It requires less computer time per iteration.
- 2-The poles are not constrained to be complex or real, but have the freedom to assume whatever configuration is best for the desired specifications.



CHAPTER VII

AN OUTLINE OF A THEORY FOR THE DESIGN OF A DIGITAL FILTER THAT WILL MEET A REQUIRED MAGNITUDE AND PHASE RESPONSE

7.1 INTRODUCTION

The method described here is that of the use of unconstrained minimization for the design of a recursive digital filter that will meet a required magnitude and phase response.

7.2 FILTER FORM

The filter form used is that of a cascade for reasons previously mentioned.

$$Y(z_1) = \prod_{k=1}^K \frac{a_k z_1^2 + b_k z_1 + c_k}{(z_1 - p_k e^{j\theta_k})(z_1 - p_k^* e^{-j\theta_k})} \prod_{j=1}^L \frac{z_1 - a_j}{z_1 - b_j} \quad (7.1)$$

7.3 PHASE RESPONSE

Let ϕ_i be the phase shift introduced by the filter at the frequency ω_1 .

$$\phi_i = \tan^{-1}(\text{Im}.Y(z_1)/\text{Re}.Y(z_1)) \quad (7.2)$$

7.4 SQUARE-ERROR CRITERION

Let Y_1^d be the desired magnitude at ω_1 , and ϕ_i^d be the desired phase at ω_1 . ω_1 is expressed in fractions of the Nyquist frequency.

Choose an error criterion of the form

$$Q(z_i, \psi) = \sum_{i=1}^M \lambda_1 (|Y(z_i, \psi)| - Y_i^d)^2 + \sum_{i=1}^M \lambda_2 (\phi_i(z_i, \psi) - \phi_i^d)^2 \quad (7.3)$$

where ψ is the vector of unknown parameters, λ_1 and λ_2 are constants ≥ 0 .

7.5 CONSTRAINTS

For stability reasons the poles of the digital filter should lie inside the unit circle of the z-plane. To take care of that we choose a new set of variables related to the variables p_k and b_j as follows

$$\begin{aligned} p_k &= \sin^2 \beta_k \\ b_j &= \sin \theta_{pj} \end{aligned} \quad (7.4)$$

Therefore,

$$Y(z_1) = \prod_{k=1}^K \frac{a_k z_1^2 + b_k z_1 + c_k}{z_1^2 - 2 \sin^2 \beta_k \cos \theta_k z_1 + \sin^4 \beta_k} \prod_{j=1}^L \frac{z_1 - a_j}{z_1 - \sin \theta_{pj}} \quad (7.5)$$

Another method of implementing constraints on the poles is that given in sec.6.1.

If we assume that we have an analog filter of the form

$$Y(s_1) = A \prod_{k=1}^K \frac{s^2 + a_k s + b_k}{s^2 + c_k^2 s + d_k^2} \quad (7.6)$$

by substituting $s = \frac{z-1}{z+1}$ in eq.(7.6) we get the required digital filter with poles constrained inside the unit circle.

$$Y(z, \psi_n) = A \prod_{k=1}^N \frac{z^2(1+a_k+b_k) + z(-2+2b_k) + (1-a_k+b_k)}{z^2(1+c_k^2+d_k^2) + z(-2+2d_k^2) + (1-c_k^2+d_k^2)} \quad (7.7)$$

7.6 TO OBTAIN $\frac{\partial Q}{\partial \psi_n}$

$$\begin{aligned} \frac{\partial Q}{\partial \psi_n} = & 2\lambda_1 \sum_{i=1}^M (|Y(z_i, \psi_n)| - Y_i^d) \frac{\partial |Y(z_i, \psi_n)|}{\partial \psi_n} + \\ & 2\lambda_2 \sum_{i=1}^M (\phi(z_i, \psi_n) - \phi_i^d) \frac{\partial \phi(z_i, \psi_n)}{\partial \psi_n} \end{aligned} \quad (7.8)$$

where n = the number of parameters .

Since , $\phi_i = \tan^{-1}(\text{Im}(Y(z_i, \psi_n))/\text{Re}(Y(z_i, \psi_n)))$

Therefore,

$$\frac{\partial \phi_i}{\partial \psi_n} = \frac{1}{|Y_i|^2} (\text{Re}(Y_i) \cdot \text{Im}(\frac{\partial Y_i}{\partial \psi_n}) - \text{Im}(Y_i) \cdot \text{Re}(\frac{\partial Y_i}{\partial \psi_n})) \quad (7.9)$$

* Note λ_1 and λ_2 are chosen such that the phase and magnitude criterions are of the same order during the minimization procedure. This is usually done in an adaptive manner, (see example at the end of this chapter).

7.7 SUMMARY OF THE SUBROUTINE FUNCT(N,X,Q,DQ)

1- Calculate Y_i $i=1,2,\dots,M$

2- Calculate ϕ_i $i=1,2,\dots,M$

3- Calculate

$$E1_i = |Y_i| - Y_i^d$$

$$E2_i = \phi_i - \phi_i^d \quad i=1,2,\dots,M$$

4- Calculate

$$Q = \lambda_1 \sum_{i=1}^M E1_i^2 + \lambda_2 \sum_{i=1}^M E2_i^2$$

5- Calculate $\frac{\partial Y_i}{\partial \psi_n}$

n=number of parameters.

6- Calculate

$$\frac{\partial Q}{\partial \psi_n} = 2\lambda_1 \sum_{i=1}^M |Y_i| \cdot \text{Re}\left(\frac{\partial Y_i}{\partial \psi_n}\right) \cdot E1_i + 2\lambda_2 \sum_{i=1}^M \frac{E2_i}{|Y_i|^2} (\text{Re}(Y_i) \frac{\partial \text{Im}(Y_i)}{\partial \psi_n} - \text{Im}(Y_i) \frac{\partial \text{Re}(Y_i)}{\partial \psi_n})$$

$$\text{Im}(Y_i) \frac{\partial \text{Re}(Y_i)}{\partial \psi_n} - \text{Re}(Y_i) \frac{\partial \text{Im}(Y_i)}{\partial \psi_n}$$

7.8 USE OF A PENALTY FUNCTION⁽⁹⁾

Another approach to this type of problem is as follows :

$$\text{minimize } Q(z_1, X) = \sum_{i=1}^M (|Y(z_1, X)| - y_1^d)^2 \quad (7.10)$$

subject to the set of constraints

$$\epsilon_1(X) = \sum_{i=1}^M \epsilon - (\phi_1^d - \phi_1) \geq 0 \quad (7.11)$$

where ϵ is the desired error in the phase.

The constraints given by eq.(7.11) could be enforced by adding a penalty function to eq.(7.10) such that if any of the constraints given by eq.(7.11) are violated, the penalty function would assume a large value, otherwise it assumes a small, or zero value.

The simplest choice of the penalty function is the function⁽⁹⁾

$$\sum_{i=1}^M |\epsilon_1(X)| - \epsilon_1(X)$$

For this choice the modified error function is

$$Q(z_1, X) = \sum_{i=1}^M (|Y(z_1, X)| - y_1^d)^2 + r \sum_{i=1}^M |\epsilon_1(X)| - \epsilon_1(X) \quad (7.12)$$

where $r > 0$, is a constant. The choice of r is critical: if it is too small the constraints will not be enforced, if too large, inaccuracies will prevent convergence.

7.9 EXAMPLE

Design a differentiator having the following specifications

$$y_1^d = w_1$$

$$\dot{y}_1^d = w_1 \times \pi$$

This problem is solved using the criterion function in sec.7.4, and the filter form is taken as that of eq.7.5.

A third order filter is considered.

ψ_0 is taken as

$$\psi_0 = (0.4, 0.4, 0.4, 0.4; 0.4, 0.4)^T$$

and $\lambda_1 = 1$, $\lambda_2 = 0.0$.

After 30 iterations, λ_2 is changed to 0.5, and 100 iterations are carried out.

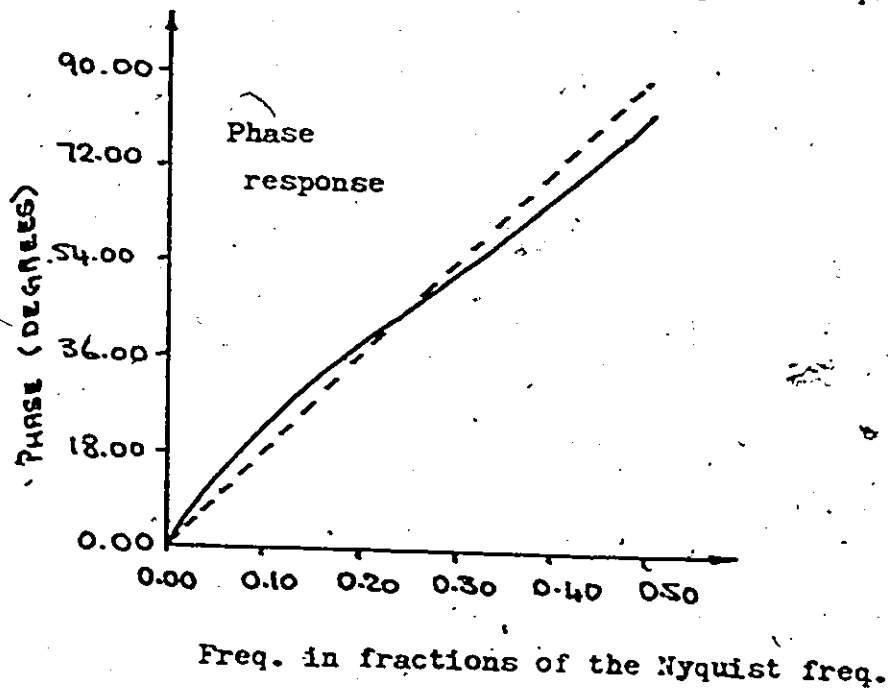
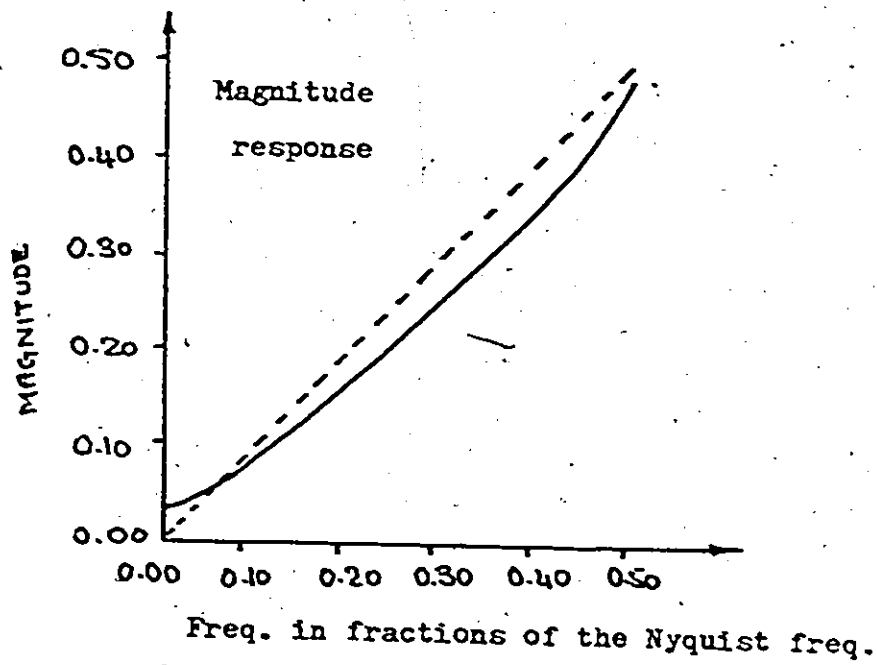
λ_2 is again changed to 0.2, and after 150 further iterations,

$$Q = 6.7776$$

$$a_1 = -0.6495$$

$$\begin{array}{l} \text{Poles } (0.10425 \times 10^{-3} \pm j0.31718 \times 10^{-4}) \\ \quad (-0.78785) \end{array}$$

$$\begin{array}{l} \text{Zeros } (0.4915 \times 10^{-1} \pm j0.49671) \\ \quad (0.67119) \end{array}$$



(Dotted curve : desired response ;
Solid curve : designed response)

CONCLUSION

In general design of digital or analog filters, there are two basic techniques available. The first is to derive an analytic expression for the required pole/zero configuration, the second is to use an iterative procedure to approach the desired configuration. It is this second approach that has been investigated in this thesis.

The work utilizes the minimization technique of Fletcher and Powell, and introduces three methods of implementing constraints on the pole-zero location of a recursive digital filter for stability and minimum phase constraint.

The three methods are :

(1) The method introduced by Steiglitz :

This method has the disadvantage that a search for the pole-zero location, to meet required magnitude specifications, is carried in the infinite z -plane, and hence for a bad choice of the starting point convergence might be very slow.

(2) The method of variable transformation :

This improves on the previous method, in the sense that the search is carried out in the unit circle where the poles and zeros are required to be located.

This method has been tested on three types of filters, and proves to give good results in a much fewer number of iterations than obtained with method (1).

Problems were solved using only a specified number of iterations, since it has been found that increasing the iterations to convergence does not significantly reduce the value of Q .

(3) The method of complex variable mapping :

This method makes use of the fact that it is easier to implement constraints on the pole-zero location in the s-plane than it is in the z-plane.

This method , though it does not converge as fast as the second method requires less computer time per iteration.

The work has been extended to include the design of continuous filters, also utilizing the Fletcher-Powell minimization technique.

Two methods were introduced for analog filter design :

(1) The variable transformation method :

This method is concerned with required magnitude specifications specified over a finite range of frequencies.

(2) The bilinear transform :

This method is a more powerful method for designing analog filters having required specifications over the infinite frequency axis.

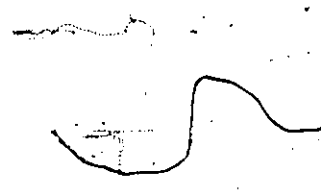
At the end of this work a method has been introduced whereby it is possible to design filters having required magnitude and phase specifications.

Further work along these lines might take into account :

(1) Arbitrary weighting of the errors at different specification points.

(2) A criterion for the choice of the multipliers λ_1 and λ_2 defined in chapter VII.

(3) It has been found that it takes fewer number of iterations to design a filter where magnitude specifications do not change abruptly. Hence, a useful investigation would be a search for a suitable transformation to transform sharp cut-off filters to a domain where the magnitude response is more smooth.



REFERENCES

- 1- K.Steiglitz , Computer-aided design of recursive digital filters , IEEE Transaction on Audio and Electroacoustics , vol.Au-18, no.2, June 1970.
- 2- B.Gold and M.Radar , Digital Processing of Signals , McGraw-Hill , 1969.
- 3- D.P.Lindorff , Theory of sampled-data control systems , John Wiley and Sons, Inc., 1965.
- 4- L.S.Lasdon, Optimization theory for large systems. Macmillan Series , 1970.
- 5- R.Fletcher and M.J.D.Powell, A rapidly convergent descent method for minimization , Computer J. , vol.6, no.2, pp.163-168, 1963.
- 6- System/360 Scientific subroutine package (360-CM-03X), version III, Programmer's manual, IBM Data Processing Division, White Plains, N.Y., Document H20-0205-3, 1968.
- 7- M.J.Box, A comparison of several current optimization methods, and the use of transformations in constrained problems, Computer J. vol.9, pp.67-77; May, 1966.
- 8- C.R.Wylie, Jr., Advanced Engineering Mathematics, McGraw-Hill, Third Edition, 1966.
- 9- F.Kuo and W.G.Magnuson, Computer oriented circuit design, Prentice-Hall, 1969.

APPENDIX

Programs and subroutines used


```

P4=DCOS(X(K1))
P5=DSIN(X(K2))
P6=DCOS(X(K2))
P7=DSINTX(K3))
P8=DCOS(X(K3))
P1=(-4.*P1+7.*P4+Z(J)+4.*(P1**2+P2**2)/71(I,J)
P2=(1.0*(1+P1**2)/71(I,J)
P3=(-4.*P5+6.*P7+Z(J)+4.*(P5**2+P6**2)/71(I,J)
P4=(2.0*(P5**2)+7)/71(I,J)
S1=S1+PL(J)*P1*PL(A1)
S2=S2+PL(J)*P2*PL(A2)
S3=S3+PL(J)*P3*PL(A3)
S4=S4+PL(J)*P4*PL(A4)
DO(I)=S1
DO(K1)=S2
DO(K2)=S3
7 DO(K3)=S4
17 I=I
IF(L.PP.0)GO TO 18
LL=4*(K+1)
L2=LL+L-1
DO 5 II=LL,L2
K2=II
K3=II+L
S1=0.0
S2=0.0
DO 1 J=10
PL(J)=2.0*(P5**2)/71(I,J)
P1=20*(S(X(K1)))/71(I,J)
P2=20*(S(X(K2)))/71(I,J)
P3=S1-PL(J)*PL(A1)
S2=S2+PL(J)*P2*PL(A2)
DO(K1)=S1
DO(K2)=S2
I=I+1
10 CONTINUE
CONTINUE

```

0001

DOUBLE PRECISION FUNCTION DREAL(X)

0002

COMPLEX*16 X,DCONJG

0003

DREAL=(X+DCONJG(X))/2.0

0004

RETURN

0005

END

SENTRY

DOUBLE PRECISION ROUTINE FOR E(X)

COMPLEX16 P,X,DC PLY,DCOMIG

P=DC PLY(C.DC,1.DC)

PIVAB=(Y-DCOMIG(X))/(2.DC)

RETURN

END

```

0001      SUBROUTINE ZPLOT(POLER,POLEI,ZEROR,ZEROI,N)
0002      LOGICAL*1 LINE,DOT/'.'/,SPACE/' '/,CROSS/'X'/,NAUGHT/'0'/,
0003      IDASH/'-'/,STROKE/'|'/
0004      DIMENSION LINE(60),POLER(1),ZEROR(1),POLEI(1),ZEROI(1)
0005      PRINT 900
0006      DO 2 J=1,59
0007      DO 3 I=1,60
0008      LINE(I)=SPACE
0009      IF(J-30)4,5,6
0010      Y=20-J
0011      IF(Y.NE.20.)GO TO 8
0012      NX=0
0013      GO TO 9
0014      NX=(Y/(TAN(ARCSIN(Y/20.))*0.6))+1.
0015      LINE(50-NX)=DOT
0016      LINE(50+NX)=DOT
0017      GO TO 10
0018      DO 7 M=1,60
0019      LINE(M)=DASH
0020      GO TO 10
0021      Y=J-20
0022      GO TO 90
0023      CONTINUE
0024      DO 20 I=1,N
0025      NPOLEI=(1.-POLEI(I))*20.+1.
0026      IF(J.NE.NPOLEI)GO TO 25
0027      NPOLER=(1.+POLER(I))*40.+1.
0028      LINE(NPOLER)=CROSS
0029      CONTINUE
0030      NZEROI=(1.-ZEROI(I))*20.+1.
0031      IF(J.NE.NZEROI)GO TO 25
0032      NZEROR=(1.+ZEROR(I))*40.+1.
0033      LINE(NZEROR)=NAUGHT
0034      CONTINUE
0035      CONTINUE
0036      PRINT 60,LINE
0037      CONTINUE
0038      60  FORMAT(20X,101A1)
0039      900  FORMAT(1H1)
0040      RETURN
0041      END

```

SUBROUTINE NAME

PURPOSE

TO FIND A LOCAL MINIMUM OF A FUNCTION OF SEVERAL VARIABLES BY THE METHOD OF FLETCHER AND POWELL

USAGE

CALL DEMP(FUNCT,N,X,F,G,EST,EPS,LIMIT,IER,H)

DESCRIPTION OF PARAMETERS

FUNCT - USER-WRITTEN SUBROUTINE CONCERNING THE FUNCTION TO BE MINIMIZED. IT MUST BE OF THE FORM
SUBROUTINE FUNCT(ARG,VAL,GRAD)

AND MUST SERVE THE FOLLOWING PURPOSE
FOR EACH N-DIMENSIONAL ARGUMENT VECTOR ARG,
FUNCTION VALUE AND GRADIENT VECTOR MUST BE COMPUTED
AND, ON RETURN, STORED IN VAL AND GRAD RESPECTIVELY.
ARG, VAL AND GRAD MUST BE OF DOUBLE PRECISION.
- NUMBER OF VARIABLES

X - VECTOR OF DIMENSION N CONTAINING THE INITIAL
ARGUMENT WHERE THE ITERATION STARTS. ON RETURN,
X HOLDS THE ARGUMENT CORRESPONDING TO THE
COMPUTED MINIMUM FUNCTION VALUE
DOUBLE PRECISION VECTOR.

F - SINGLE VARIABLE CONTAINING THE MINIMUM FUNCTION
VALUE ON RETURN, I.E. F=FUNCT(X).
DOUBLE PRECISION VARIABLE.

G - VECTOR OF DIMENSION N CONTAINING THE GRADIENT
VECTOR CORRESPONDING TO THE MINIMUM ON RETURN.
I.E. G=GRAD(X).
DOUBLE PRECISION VECTOR.

EST - IS AN ESTIMATE OF THE MINIMUM FUNCTION VALUE.
SINGLE PRECISION VARIABLE.

EPS - TEST VALUE REPRESENTING THE EXPECTED ABSOLUTE ERROR.
A REASONABLE CHOICE IS $10^{-(14)}$, I.E.
SLIGHTLY GREATER THAN $10^{-(14)}$, WHERE 14 IS THE
NUMBER OF SIGNIFICANT DIGITS IN FLOATING POINT
REPRESENTATION.
SINGLE PRECISION VARIABLE.

LIMIT - MAXIMUM NUMBER OF ITERATIONS.

IER - ERROR PARAMETER

IER = 0 MEANS CONVERGENCE WAS OBTAINED

IER = 1 MEANS NO CONVERGENCE IN LIMIT ITERATIONS

IER = -1 MEANS ERRORS IN GRADIENT CALCULATION

IER = 2 MEANS DIRECT SEARCH TECHNIQUE INDICATES
IT IS LIKELY THAT THERE EXISTS NO MINIMUM.

H - WORKING STORAGE OF DIMENSION $2 \times (N+7)/2$.
DOUBLE PRECISION ARRAY.

REMARKS

- (I) THE SUBROUTINE NAME REPLACING THE DUMMY ARGUMENT, FUNCT
MUST BE DECLARED AS EXTERNAL IN THE CALLING PROGRAM.
- (II) IER IS SET TO 2 IF, STOPPING IN ONE OF THE COMPUTED
DIRECTIONS, THE FUNCTION WILL NEVER INCREASE WITHIN
A TOLERABLE RANGE OF ARGUMENT.

IER = 2 MAY OCCUR ALSO IF THE INTERVAL WHERE F INCREASES IS SMALL AND THE INITIAL ARGUMENT WAS RELATIVELY FAR AWAY FROM THE MINIMUM SUCH THAT THE MINIMUM WAS OVERLEAPED. THIS IS QUOTE THE SEARCH TECHNIQUE WHICH DOWNSIZES THE STEPSIZE UNTIL A POINT IS FOUND WHERE THE FUNCTION INCREASES.

SUBROUTINES AND FUNCTION SUBPROGRAMS REQUIRED
FUNCT

METHOD

THE METHOD IS DESCRIBED IN THE FOLLOWING ARTICLE
R. FLETCHER AND W.J.D. POWELL, A RAPID DESCENT METHOD FOR MINIMIZATION,
COMPUTER JOURNAL VOL.6, ISS. 2, 1963, PP.163-169.

0001

SUBROUTINE DEFW(FUNCT,N,X,F,G,EST,FES,LIMIT,IER,H)

0002

DIMENSIONED QUANTY VARIABLES

0003

DIMENSION H(1),X(1),G(1)

DOUBLE PRECISION X,F,FX,FY,GRD,HARDY,GRDY,F,G,GX,GY,ALFA,DLF1,
IAMBDA,I,Z,K

0004

COMPUTE FUNCTION VALUE AND GRADIENT VECTOR FOR INITIAL ARGUMENT
CALL FUNCT(N,X,F,G)

0005

RESET ITERATION COUNTER AND GENERATE IDENTITY MATRIX

0006

IER=0

0007

K=1

0008

NR=N+1

0009

NR=N+1

0010

1 K=NR

0011

DO 4 J=1,N

0012

H(K)=1.00

0013

NJ=N-J

0014

IF(NJ)5,6,2

0015

2 DO 2 I=1,NJ

0016

ALFA=L

0017

3 H(KI)=0.00

0018

4 K=KI+1

0019

START ITERATION LOOP
5 CONTINUE(1)

0020

SAVE FUNCTION VALUE, ARGUMENT VECTORS AND GRADIENT VECTORS

0021

DLDF=F

0022

DO 6 J=1,N

0023

K=N+J

0024

H(K)=G(J)

0025

X=X+H

H(K)=X(J)

0026

DETERMINE DIRECTION VECTOR H

0027

K=J+NR

I=0.00

```

0028      DO 2 L=1,N
0029      T=T-G(L)*H(K)
0030      IF (L-J) 6,7,7
0031      K=K+1
0032      IF (L-J) 3
0033      K=K+1
0034      CONTINUE
0035      H(J)=T

C
C      CHECK WHETHER FUNCTION WILL DECREASE STEPPING ALONG H.
0036      DY=0.00
0037      HTRY=0.00
0038      GTRY=0.00

C
C      CALCULATE DIRECTIONAL DERIVATIVE AND TEST VALUES FOR DIRECTION
C      VECTOR H AND GRADIENT VECTOR G.
0039      DO 10 J=1,N
0040      HTRY=HTRY+DAYS(H(J))
0041      GTRY=GTRY+DAYS(G(J))
0042      10 DY=DY+H(J)*G(J)

C
C      REPEAT SEARCH IN DIRECTION OF STEEPEST DESCENT IF DIRECTIONAL
C      DERIVATIVE APPEARS TO BE POSITIVE OR ZERO.
0043      IF (DY) 11,51,51

C
C      REPEAT SEARCH IN DIRECTION OF STEEPEST DESCENT IF DIRECTION
C      VECTOR H IS SMALL COMPARED TO GRADIENT VECTOR G.
0044      11 IF (HTRY/GTRY) 12,51,51,12

C
C      SEARCH MINIMUM ALONG DIRECTION H
C
C      SEARCH ALONG H FOR POSITIVE DIRECTIONAL DERIVATIVE
0045      12 FY=F
0046      ALFA=2.00*(F1-F)/DY
0047      AMROA=1.00

C
C      USE ESTIMATE FOR STEPSIZE ONLY IF IT IS POSITIVE AND LESS THAN
C      1. OTHERWISE TAKE 1. AS STEPSIZE
0048      IF (ALFA) 13,15,13
0049      13 IF (ALFA-AMROA) 14,15,15
0050      14 AMROA=ALFA
0051      15 AMFA=0.00

C
C      SAVE FUNCTION AND DERIVATIVE VALUES FOR OLD ARGUMENT
0052      16 FX=FY
0053      DY=DY

C
C      STEP ARGUMENT ALONG H
0054      DO 17 I=1,N
0055      17 X(I)=X(I)+AMROA*H(I)

C
C      COMPUTE FUNCTION VALUE AND GRADIENT FOR NEW ARGUMENT
0056      CALL FUNCT(X,F,G)
0057      FY=F

C
C      COMPUTE DIRECTIONAL DERIVATIVE DY FOR NEW ARGUMENT. TERMINATE
C      SEARCH IF DY IS POSITIVE. IF DY IS ZERO THE MINIMUM IS FOUND
0058      DY=0.00

```

20/51/7

```
DO 15 I=1,N
  DY=DY+G(I)*H(I)
  IF(DY)15,36,22.
```

15 IF (FY-FX)25,22,22

REPEAT SEARCH AND DOUBLE STEPSIZE FOR FURTHER SEARCHES
20. AMCOS=AMCDA+ALFA

END OF SEARCH LOOP

TERMINATE IF THE CHANGE IN ARGUMENT GETS VERY LARGE
IF (HNSNDRAMBA-1.01) 16,16,21

```

21 IF I=2:
    RETURN
    LINEAR SEARCH TECHNIQUE INDICATES THAT NO MINIMUM EXISTS

```

INTERPOLATE CUBICALLY IN THE INTERVAL OBTAINED BY THE SEARCH ABOVE AND COMPUTE THE ARGUMENT ψ FOR WHICH THE INTERPOLATION POLYNOMIAL IS MINIMIZED

```

22 T=C.DC
23 IF (AMPERA) 24, 26, 24
24 Z=P.RC*(FY-FX)/1200A+BY+BY
    ALFA=DNAX1(DARS(Z),DARS(FX),DARS(FY))
    SILEA=Z/ALFA

```

~~DATE=7/7/84~~

```

IF (ALFA) 250,251,250

```

$$25) \text{ ALF} = (DY - 7 + Y) / \text{ALF}$$
$$251 \quad \text{ALSA} = (7 + 0Y - 6) / (0Z + 0X + 7 + 0V)$$

252 $A \setminus B = A \setminus (A \cap B)$

1970 20 12 12

$$26. \quad X(1) = Y(1) + (T-1)(\bar{Y} - Y(1))$$

TERMINATE, IF THE VALUE OF THE ACTUAL FUNCTION AT x IS LESS THAN APPROXIMATE VALUES AT THE INTERVAL ENDS. OTHERWISE, FIND THE INTERVAL BY CHOOSING ONE END-POINT EQUAL TO x AND REPEAT THE INTERPOLATION. WHICH END-POINT IS CHOSEN DEPENDS ON THE VALUE OF THE FUNCTION AND ITS GRADIENT AT x .

CALL ELACT(1,X,F,G)
IF(F-EX)27,27,29

17 (1-17) 30, 20, 20

29 DALFA=C.IV

100, 200 1=1, 2

29 CALFA = CALFA + G(I) * H(I)

IF (CALFA) 22, 22, 32

30 IF (F-FX) 32, 31, 23

71 IF (DX-DALFA) 32, 34, 32

32 $F_X = \Gamma$

$\mathcal{O}X = \mathcal{O}A_1$

```

0006      AMBDA=ALFA
0007      GO TO 23
0008      22 IF(EV-F)25,34,35
0009      24 IF(DY-DALFA)35,34,35
0100      35 Y=F
0101      DY=DALFA
0102      AMBDA=AMBDA-ALFA
0103      GO TO 22
C
C      TERMINATE, IF FUNCTION HAS NOT DECREASED DURING LAST ITERATION
0104      36 IF(OLD-F+EPS)51,37,38
C
C      COMPUTE DIFFERENCE VECTORS OF ARGUMENT AND GRADIENT FROM
C      TWO CONSECUTIVE ITERATIONS
0105      39 DO 37 J=1,N
0106      K=X+J
0107      P(K)=G(J)-H(K)
0108      K=N+K
0109      37 H(K)=X(J)-H(K)
C
C      TEST LENGTH OF ARGUMENT DIFFERENCE VECTOR AND DIRECTION VECTOR
C      IF AT LEAST N ITERATIONS HAVE BEEN EXECUTED, TERMINATE, IF
C      BOTH ARE LESS THAN EPS
0110      IF=0
0111      IF(KOUNT-N)42,39,39
0112      T=0.00
0113      Z=1.00
0114      DO 40 J=1,N
0115      K=X+J
0116      L=H(K)
0117      K=N+K
0118      T=T+ABS(H(K))
0119      Z=Z+H(K)
0120      IF(ABS(COS) )1,41,42
0121      41 IF(T-EPS)53,56,42
C
C      TERMINATE, IF NUMBER OF ITERATIONS WOULD EXCEED LIMIT
0122      42 IF(KOUNT-LIMIT)43,54,54
C
C      REPEAT UPDATING OF MATRIX H
0123      43 ALFA=1.00
0124      DO 47 J=1,N
0125      K=J+N
0126      W=0.00
0127      DO 44 L=1,N
0128      K1=N+L
0129      W=W+H(K1)*H(K)
0130      IF(L-J)44,45,45
0131      44 K=X+K1-L
0132      GO TO 44
0133      45 K=X+1
0134      45 CONTINUE
0135      K=N+J
0136      ALFA=ALFA+W*H(K)
0137      47 H(J)=W
C
C      REPEAT SEARCH IN DIRECTION OF STEEPEST DESCENT IF RESULTS
C      ARE NOT SATISFACTORY

```

```

0138      IF(Z#2)F214R,1,4R
      C
      C      UPDATE MATRIX H
      C      40 K=1
0139      40 K=1
0140      40 40 I=1,3
0141      K1=N2+1
0142      40 40 J=1,3
0143      NJ=N2+J
0144      H(K)=H(K)+H(KL)*H(NJ)/Z-H(L)*H(J)/ALFA
0145      50 K=K+1
0146      GO TO 5A
      C
      C      END OF ITERATION LOOP
      C
      C      NO CONVERGENCE AFTER LIMIT ITERATIONS
0147      50 H2=1
0148      GOTO 1
      C
      C      RESTORE OLD VALUES OF FUNCTION AND ARGUMENTS
0149      51 40 50 J=1,3
0150      K=N2+1
0151      52 X(J)=H(K)
0152      CALL F10(X,Y,F,G)
      C
      C      REPEAT SEARCH IN DIRECTION OF STEEPEST DESCENT IF DERIVATIVE
0153      FAILS TO BE SUFFICIENTLY SMALL
      C      IF(GABS-ERR)55,55,50
      C
      C      TEST FOR REPEATED FAILURE OF ITERATION
0154      53 IF(ABS(F1,F2,F3)
0155      54 J=1
0156      55 50 1
0157      56 1
0158      57 1
0159      58 1
0160      59 1
      C

```


114

C. A. B. 100, 100, 100.

DV=3.14159102300

.....


```

24  FORMAT(1H1, 'PLOTTING OF THE GIVEN FILTER MAGNITUDE RESPONSE')
    CALL PLOT3(YS,YDS,I)
    WRITE (6,23)
28  FORMAT(1H1, 'PLOTTING OF THE GIVEN FILTER PHASE RESPONSE')
    CALL PLOT3(US,PHASES,I)
    DO 100 I=1,N
29  US(I)=0.5*(Y(I)+Y(N+1-I))
    WRITE (6,72)
72  FORMAT(4X,11,20X,Y(I),30X,US(I),30X,PHASE(I),10X)
    WRITE (6,73)(I,Y(I),US(I),PHASE(I),I=1,N)
73  FORMAT(4X,13,4X,D20.12,21X,D20.12,20X,D20.12)
    DO 100 I=1,N
    PHASES(I)=SGL(PHASE(I))
21  YS(I)=SGL(Y(I))
    WRITE (6,24)
24  FORMAT(1H1, 'PLOTING OF THE DESIGNED FILTER MAGNITUDE RESPONSE')
    CALL PLOT3(YS,YS,I)
    WRITE (6,25)
25  FORMAT(1H1, 'PLOTING OF THE DESIGNED FILTER PHASE RESPONSE')
    CALL PLOT3(US,PHASES,I)
    DO 100 I=1,N
    N1=N+1-I
    POLES(I)=SGL(MAGL(POLES1(I)))
    POLES(N1)=SGL(MAGL(POLES2(I)))
    POLES(I)=SGL(POLES(I)+POLES(N1))
    POLES(N1)=SGL(POLES(N1)+POLES(I))
    ZEROS(I)=SGL(MAGL(ZEROS1(I)))
    ZEROS(N1)=SGL(MAGL(ZEROS2(I)))
    ZEROS(I)=SGL(ZEROS(I)+ZEROS(N1))
    ZEROS(N1)=SGL(ZEROS(N1)+ZEROS(I))
100  I=I+1
    LL=2*N+1
    L2=LL-I-1
    DO 100 I=LL,L2
    POLES(I)=SGL(POJ(I))
    POLES(I+1)=0.0
    ZEROS(I)=SGL(ZOJ(I))
    ZEROS(I+1)=0.0
100  I=I+1
    NP=2*N+1
    CALL ZPLOT(POLES,POLES1,ZEROS,ZEROS1,NP)
    C1=0.00
    C2=0.00
    DO 500 I=1,N
    C1=C1+F1(I)**2
    C2=C2+F2(I)**2
500  WRITE (6,600) C1,C2
600  FORMAT(3X, 'SQUARE ERROR IN MAGNITUDE=',D20.12/3X, 'SQUARE ERROR
    IN PHASE=',D20.12)
    PUNCH 4000, (YS(I),US(I),PHASES(I),I=1,N)
400  FORMAT(3F12.5)
    STOP
    END

```

SUBROUTINE FUNCT(,X,Y,Z)

C... DIMENSIONING... PHASE(30), PHASE(30), YD(30), (30), F1(30),
 C... F2(30), PL1(30), PL2(30), C(30), AI(30), R(30), Z(30), H(30), RINK, 7, S2(Y, 30), S7(L,
 C

DOUBLE PRECISION X(30), Y(30), Z(30), PHASE(30), PHASE(30), YD(30), (30),
 1 PY, F1(30), F2(30), S1, S2, S3, S4, S5, F1(30), PL2(30), AI(30), C(30),
 1, S6, S7, S8, S9, S10, S11, S12, S13, S14, S15, S16, S17, S18, S19, S20,
 COMPLEX*16 Z(30), F(30), F1(2,30), F2(2,30), S(2,30), S7(1,30), S8(1,30),
 1 S1, S2, S3, S4, S5, COMPLEX*16
 COMMON /71/7, H

COMMON /82/ PHASE, PLASD, YD, H, PY, F1, F2
 COMMON /83/ ALA1, D1, ALA2, D2
 COMMON /74/ A, L

O=C, D3

DO 2 J=1,

PL1=PL1(1, J), F1(1, J)

DO 3 I=1,

F1=F1+I

K2=2*K+1

K3=2*K+1

K4=2*K+1

K5=2*K+1

K6=2*K+1

K7=2*K+1

K8=2*K+1

K9=2*K+1

K10=2*K+1

K11=2*K+1

K12=2*K+1

K13=2*K+1

K14=2*K+1

K15=2*K+1

K16=2*K+1

K17=2*K+1

K18=2*K+1

K19=2*K+1

K20=2*K+1

K21=2*K+1

K22=2*K+1

K23=2*K+1

K24=2*K+1

K25=2*K+1

K26=2*K+1

K27=2*K+1

K28=2*K+1

K29=2*K+1

K30=2*K+1

K31=2*K+1

K32=2*K+1

K33=2*K+1

K34=2*K+1

K35=2*K+1

K36=2*K+1

K37=2*K+1

K38=2*K+1

K39=2*K+1

K40=2*K+1

```

A1=7(J)**2/R1(I,J)
A2=7(J)/R1(I,J)
A3=1.0/R1(I,J)
A4=(4.0*DSIN(X(K3))*DCOS(X(K3))*DCOS(X(K4))*Z(J)-4.0*(DSIN(X(K3))
J ***3)*DCOS(X(K3)))/R2(I,J)
A5=-2.0*(DSIN(X(K3))*2)*DSIN(X(K4))*7(J)/R2(I,J)
A6=(1(J)*T1(A2*H(J))-A1(J)*DREAL(A2*H(J)))/C(J)
A7=(R(J)*DI*AG(A2*H(J))-A1(J)*DREAL(A2*H(J)))/C(J)
A8=(R(J)*DI*AG(A3*H(J))-A1(J)*DREAL(A3*H(J)))/C(J)
A9=(R(J)*DI*AG(A4*H(J))-A1(J)*DREAL(A4*H(J)))/C(J)
A10=(R(J)*DI*AG(A5*H(J))-A1(J)*DREAL(A5*H(J)))/C(J)
S1=S1+PL1(J)*DREAL(A1)+PL2(J)*A6
S2=S2+PL1(J)*DREAL(A2)+PL2(J)*A7
S3=S3+PL1(J)*DREAL(A3)+PL2(J)*A8
S4=S4+PL1(J)*DREAL(A4)+PL2(J)*A9
2 S5=S5+PL1(J)*DREAL(A5)+PL2(J)*A10
DO(I)=S1
DO(K1)=S2
DO(K2)=S3
DO(K3)=S4
7 DO(K4)=S5
I=1
LL=5*K+1
L2=LL+L-1
DO 10 II=LL,L2
K5=II
K4=II+L
S1=0.00
S2=0.00
DO 10 J=1,10
A1=-1.07/7(I,J)
A2=DCOS(X(K5))/R1(I,J)
A3=(1(J)*T1(A2*H(J))-A1(J)*DREAL(A2*H(J)))/C(J)
A4=(1(J)*T1(A3*H(J))-A1(J)*DREAL(A3*H(J)))/C(J)
S1=S1+PL1(J)*DREAL(A1)+PL2(J)*A3
C S2=S2+PL1(J)*DREAL(A2)+PL2(J)*A4
DO(K5)=S1
DO(K6)=S2
10 I=I+1
RETURN
END

```

C..DESIGN OF AN ANALOG FILTER USING BILINEAR TRANSFORMATION OF A DIGIT
..... AT 100000

DIMENSION YS(30), XS(30), CMAS(30), YDS(30)

DOUBLE PRECISION X(16), Y(16), YD(31), W(31), WC(164), C, A(3), Z(31),

1 C(2), D(3), AJ(2), BJ(2), CS(1), CY, Y(31), CDS, CMAS, 10
1 ,OPAL,DTAT,ATAG,BATV,ILND,CMASE(31)

1 ,S1,S2,SUM1,SUM2,CA(3),CB(3),CC(3),AJ(2),BJ(2),AA(3),BAC

COMPLEX#16 Z(31),CP,CC,CLX,CDEY2,W(31),Z1,Z2,POLE1(3),POLE2(3)

1 ,ZSP01(3),ZSP02(3),P,PCOMD,CDSCOT,S(31)

COMMON/P1/Z,P

COMMON/P2/YD,AD

COMMON/P3/I,K,L

PY=3.141592653589793

EXTERNAL FNCT

C

C

C..DIMENSIONING X(16),Y(16),YD(31),W(31),WC(164),C,A(3),Z(31),
C AJ(2),BJ(2),Z(31),POLE1(3),POLE2(3),ZSP01(3),ZSP02(3),
C..DIMENSIONING CA(3),CB(3),CC(3),AJ(2),BJ(2),AA(3),BAC

C

C..K,K,PL ARE AS SPECIFIED IN THE LITERATURE. IS THE NUMBER OF
C..THIS FILTER SPECIFICATIONS ARE TO BE GIVEN BETWEEN THE DOTTED LI

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C

C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF
C..THIS IS THE FIRST LINE OF THE PROGRAM. IS THE NUMBER OF

C. CONDITIONS AFTER THE SELECTED FREQUENCIES SHOULD BE DONE I
 C. OF 1.0, BETWEEN 0.5 AND 1.0, WHERE W(ND)=0.5, AND V(I)=1.0 IS IN E.C
 C. FREQUENCY IN THE S-PLANE
 C. DIGITAL FREQ. AXIS 0.0-----0.5-----1.0
 C. ANALOG FREQ. AXIS 0.0-----INFINITY
 C. SCALING

 C. SCALING FREQUENCIES AND MAPPING
 DO 3 I=1,N
 V(I)=W(I)*ALMID
 W(I)=2.0*DATA(W(I))/PY

DO 5 I=1,N
 W(I+1)=W(I)+0.05
 5 V(I+1)=1.0
 W=70
 LIMIT=30

 C. SCALING FREQUENCIES AND MAPPING
 DO 4 I=1,N
 W(I)=W(I)*ALMID
 4 Z(I)=CONJUGATE(W(I))
 EST=1.1
 EST=1.0/10.0
 CALL SCALING (Z, W, EST, EST, 1.0, 1.0, 1.0, 1.0)
 4 W(I)=W(I)*EST

DO 6 I=1,N
 W(I)=W(I)*EST
 6 W(I)=W(I)*EST
 100 FOR I=1,N
 W(I)=W(I)*EST
 W1=1.0
 W2=2.0
 W3=3.0
 W4=-0.5
 W5=0.5
 W6=1.0
 W7=-1.0
 W8=1.0
 W9=-1.0
 W10=1.0
 W11=-1.0
 W12=1.0
 W13=-1.0
 W14=1.0
 W15=-1.0
 W16=1.0
 W17=-1.0
 W18=1.0
 W19=-1.0
 W20=1.0
 W21=-1.0
 W22=1.0
 W23=-1.0
 W24=1.0
 W25=-1.0
 W26=1.0
 W27=-1.0
 W28=1.0
 W29=-1.0
 W30=1.0
 W31=-1.0
 W32=1.0
 W33=-1.0
 W34=1.0
 W35=-1.0
 W36=1.0
 W37=-1.0
 W38=1.0
 W39=-1.0
 W40=1.0
 W41=-1.0
 W42=1.0
 W43=-1.0
 W44=1.0
 W45=-1.0
 W46=1.0
 W47=-1.0
 W48=1.0
 W49=-1.0
 W50=1.0
 W51=-1.0
 W52=1.0
 W53=-1.0
 W54=1.0
 W55=-1.0
 W56=1.0
 W57=-1.0
 W58=1.0
 W59=-1.0
 W60=1.0
 W61=-1.0
 W62=1.0
 W63=-1.0
 W64=1.0
 W65=-1.0
 W66=1.0
 W67=-1.0
 W68=1.0
 W69=-1.0
 W70=1.0
 W71=-1.0
 W72=1.0
 W73=-1.0
 W74=1.0
 W75=-1.0
 W76=1.0
 W77=-1.0
 W78=1.0
 W79=-1.0
 W80=1.0
 W81=-1.0
 W82=1.0
 W83=-1.0
 W84=1.0
 W85=-1.0
 W86=1.0
 W87=-1.0
 W88=1.0
 W89=-1.0
 W90=1.0
 W91=-1.0
 W92=1.0
 W93=-1.0
 W94=1.0
 W95=-1.0
 W96=1.0
 W97=-1.0
 W98=1.0
 W99=-1.0
 W100=1.0

DO 7 I=1,N
 W(I)=W(I)*EST
 7 W(I)=W(I)*EST
 100 FOR I=1,N
 W(I)=W(I)*EST
 W1=1.0
 W2=2.0
 W3=3.0
 W4=-0.5
 W5=0.5
 W6=1.0
 W7=-1.0
 W8=1.0
 W9=-1.0
 W10=1.0
 W11=-1.0
 W12=1.0
 W13=-1.0
 W14=1.0
 W15=-1.0
 W16=1.0
 W17=-1.0
 W18=1.0
 W19=-1.0
 W20=1.0
 W21=-1.0
 W22=1.0
 W23=-1.0
 W24=1.0
 W25=-1.0
 W26=1.0
 W27=-1.0
 W28=1.0
 W29=-1.0
 W30=1.0
 W31=-1.0
 W32=1.0
 W33=-1.0
 W34=1.0
 W35=-1.0
 W36=1.0
 W37=-1.0
 W38=1.0
 W39=-1.0
 W40=1.0
 W41=-1.0
 W42=1.0
 W43=-1.0
 W44=1.0
 W45=-1.0
 W46=1.0
 W47=-1.0
 W48=1.0
 W49=-1.0
 W50=1.0
 W51=-1.0
 W52=1.0
 W53=-1.0
 W54=1.0
 W55=-1.0
 W56=1.0
 W57=-1.0
 W58=1.0
 W59=-1.0
 W60=1.0
 W61=-1.0
 W62=1.0
 W63=-1.0
 W64=1.0
 W65=-1.0
 W66=1.0
 W67=-1.0
 W68=1.0
 W69=-1.0
 W70=1.0
 W71=-1.0
 W72=1.0
 W73=-1.0
 W74=1.0
 W75=-1.0
 W76=1.0
 W77=-1.0
 W78=1.0
 W79=-1.0
 W80=1.0
 W81=-1.0
 W82=1.0
 W83=-1.0
 W84=1.0
 W85=-1.0
 W86=1.0
 W87=-1.0
 W88=1.0
 W89=-1.0
 W90=1.0
 W91=-1.0
 W92=1.0
 W93=-1.0
 W94=1.0
 W95=-1.0
 W96=1.0
 W97=-1.0
 W98=1.0
 W99=-1.0
 W100=1.0

```

CALL PLOT3(YS,YS,I)
WRITE(6,11) AAD
DO 300 I=1,K
  PHAS(I)=DATA(DIAG(X(I))/DREAL(S(I)))
  PHAS(I)=SIN(PI*PHAS(I))
  DO 100 J=1,L
    X1=X(I)+Y(J)*X(I)*X(I)+Y(J)*Y(J)*X(I)*X(I)
    X2=X(I)+Y(J)*X(I)*X(I)+Y(J)*Y(J)*X(I)*X(I)
    POLF1(I)=-CA(I)/2.0+CDSCOT(31)/2.0
    POLF2(I)=-CA(I)/2.0+CDSCOT(31)/2.0
    ZFPO1(I)=-CA(I)/2.0+CDSCOT(32)/2.0
    ZFPO2(I)=-CA(I)/2.0+CDSCOT(32)/2.0
    WRITE(6,15) I,POLF1(I)
    WRITE(6,16) I,POLF2(I)
    WRITE(6,17) I,ZFPO1(I)
    WRITE(6,18) I,ZFPO2(I)
  DO 100 J=1,L
    WRITE(6,20) I, XJ(I)
    WRITE(6,21) I, YJ(I)
    WRITE(6,200)
    ECORAT(AX,11,20X,1Y(I),30X,1Y(I))
    WRITE(6,201) I,YS(I),YS(I),I=1,K
    ECORAT(10X,10,10X,112.5,20X,112.5)
    PRINT 400,(Y(I),X(I),PHAS(I),Y(I),I=1,K)
  DO 400 ECORAT(4F12.5)
  STOP
END

```



```

WRITE(6,14)C
14  FORMAT(5X,'0=',D20.12)
DO 23 I=1,K
K1=K+I
K2=2*K+I
K3=3*K+I
P(I)=X(I)
R(I)=X(K1)
C(I)=X(K2)
D(I)=X(K3)
R1=DCPLX(C(I)**2-4.0*D(I),0.00)
R2=DCPLX(R(I)**2-4.0*D(I),0.00)
POLF1(I)=(-C(I)+COSQRT(R1))/2.0
POLF2(I)=(-C(I)-COSQRT(R1))/2.0
P1(I)=CDABS(POLF1(I))
P2(I)=CDABS(POLF2(I))
ZFPO1(I)=(-A(I)+COSQRT(R2))/2.0
ZFPO2(I)=(-A(I)-COSQRT(R2))/2.0
Z1(I)=CDABS(ZFPO1(I))
Z2(I)=CDABS(ZFPO2(I))
IF(P1(I)-1.00)13,15,16
16 POLF1(I)=ANGLE(POLF1(I))/P1(I)
15 IF(P2(I)-1.00)17,17,17
17 POLF2(I)=ANGLE(POLF2(I))/P2(I)
17 IF(Z1(I)-1.00)19,19,20
20 ZFPO1(I)=ANGLE(ZFPO1(I))/Z1(I)
19 IF(Z2(I)-1.00)21,21,22
22 ZFPO2(I)=ANGLE(ZFPO2(I))/Z2(I)
21 Z(I)=- (ZFPO1(I)+ZFPO2(I))
C(I)=ZFPO1(I)+ZFPO2(I)
C(I)=-(POLF1(I)+POLF2(I))
D(I)=POLF1(I)*POLF2(I)
X(K1)=C(I)
X(K2)=C(I)
X(K3)=D(I)
23 CONTINUE
ITER=ITER+1
IF(ITER.GT.1)GO TO 202
DO 100 I=1,K
IF(Z1(I).GT.1.00) GO TO 201
IF(Z2(I).GT.1.00) GO TO 201
IF(P1(I).GT.1.00) GO TO 201
IF(P2(I).GT.1.00) GO TO 201
100 CONTINUE
202 CONTINUE
CALL FUNCT(I,X,C,D)
WRITE(6,12) AD
12  FORMAT(5X,'120=',D20.12)
DO 24 I=1,K
24  WRITE(6,25) I,POLF1(I),I,POLF2(I),I,ZFPO1(I),I,ZFPO2(I)
25  FORMAT(5X,'POLF1(',I3,')=',D20.12,4X,D20.12/5X,'POLF2(',I3,')=',
1  D20.12,4X,D20.12/5X,'ZFPO1(',I3,')=',D20.12,4X,D20.12/
1  5X,'ZFPO2(',I3,')=',D20.12,4X,D20.12)
DO 23 I=1,K
23  WRITE(6,32) I,A(I),I,Z(I),I,C(I),I,D(I)
32  FORMAT(5X,'A(',I3,')=',D20.12/5X,'B(',I3,')=',D20.12/5X,'C(',I3,
1  ')=',D20.12/5X,'D(',I3,')=',D20.12)
DO 26 I=1,K
Y(I)=40*CDABS(P(I))
ARG=ATN(REAL(P(I))/REAL(P(I)))

```

```

26  PHASE(I)=DATAN(ARG)*180./PI
    WRITE(6,27)
27  FORMAT(5X,'I',20X,'Y(I)',20X,'M(I)',22X,'PHASE(I)')
    WRITE(6,28)(I,Y(I),M(I),PHASE(I),I=1,N)
28  FORMAT(4X,I3,9X,D20.12,7X,D20.12,10X,D20.12)
    DO 29 I=1,N
        YS(I)=SINGL(Y(I))
        MS(I)=SINGL(M(I))
        YS(I)=SINGL(Y(I))
29  PHASE(I)=SINGL(PHASE(I))
    WRITE(6,30)
30  FORMAT(1H1,'PLOTING OF THE GIVEN FILTER MAGNITUDE RESPONSE')
    CALL PLOT3(MS,YS,N)
    WRITE(6,31)
31  FORMAT(1H1,'PLOTING OF THE DESIGNED FILTER MAGNITUDE RESPONSE')
    CALL PLOT3(MS,YS,N)
    WRITE(6,101)
101  FORMAT(1H1,'PLOTING OF THE FILTER PHASE RESPONSE')
    CALL PLOT3(MS,PHAS,N)
    DO 500 I=1,K
        K1=K+I
        POLE1(I)=SINGL(DREAL(POLE1(I)))
        POLE1(K1)=SINGL(DREAL(POLE2(I)))
        POLE1(I)=SINGL(AI(POLE1(I)))
        POLE1(K1)=SINGL(AI(POLE2(I)))
        ZERO1(I)=SINGL(DREAL(ZERO1(I)))
        ZERO1(K1)=SINGL(DREAL(ZERO2(I)))
        ZERO1(I)=SINGL(AI(ZERO1(I)))
        ZERO1(K1)=SINGL(AI(ZERO2(I)))
500  N=2*N
    CALL ZPLOT(POLE1,POLE1,ZERO1,ZERO1,N)
    N=N+1
    LIMIT=100
    X(5)=X(4)
    X(7)=X(4)
    X(6)=X(3)
    X(5)=X(2)
    X(4)=X(2)
    X(3)=X(2)
    X(2)=X(1)
    IF(X.EQ.2) GO TO 201
    STOP
    END

```

COUNTRY

SUBROUTINE SUBCT($\frac{P}{X}$, ...)

COMMON /PRECISION/ N(1), C, PC(1), C2(2,30), SUM2, F(30), S1, S2, S3, S4, =

1 YP(30), CPAL, CPAS

CC PLX=18, P, C1(2,30), C2(2,30), X(30), Z(30), DO PLX, =1, 42, 43, =

CC C1(1)/C1/2, 1

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

CC C1(1)/C1/2, 43

CC C2(1)/C2/2, 43

SENTRY

```

COMPLEX FUNCTION ANGLE*16(X)
  DOUBLE PRECISION THETA, DATA, AI, DREAL, DSI, DCIS
  COMPLEX*16 X, DCOMPLX
  THETA=DATAN(DI(X)/DREAL(X))
  ANGLE=DCOMPLX(DCOS(THETA),DSIN(THETA))
  RETURN
END

```

```

DOUBLE PRECISION FUNCTION AIM(X)
  COMPLEX*16 X,P,DCOMPLX
  P=DCOMPLX(1.0,1.0)
  AIM=(X-DCOMPLX(P))/(1.0-P)
  RETURN
END

```

KENTRY

DESIGN OF DIGITAL FILTERS BY IMPLEMENTING CONSTRAINTS ON AN ANALOG

```

1  Y(30),HC(30),POLE(4),POLE1(4),ZEROS(4),ZEROS1(4)
2  COMPLEX Z(30),CP,R1,R2,POLE1(2),POLE2(2),ZEROS1(2),ZEROS2(2),H(30)
3  C=PLX,CFXP,CSORT
4  COMMON/31/7,H
5  COMMON/32/AR,VR
6  COMMON/33/74,H
7  COMMON/32/RRRNT
8  EXTERNAL FUNCT

```

...FILTER SPECIFICATIONS

RY=2.141592

K=1

N=27

X(1)=1.414

X(2)=1.0

X(3)=1.0

X(4)=1.0

Y(1)=1.0

Y(2)=0.

DO 3 I=1,12

Y(1)=1.

3 Y(11)=Y(1)+0.01

Y(11)=0.0

Y(12)=0.0

DO 12 I=12,21

Y(1)=0.

12 Y(11)=Y(1)+0.01

DO 12 I=22,27

Y(1)=0.

15 Y(1)=Y(1)+0.1

...EVALUATION OF Z(I)

DO 100 I=1,

CP=C*PLX(C,H,X(1)+CPY)

100 Z(I)=CP*Z(I)

EST=0.1

ERR=1.0/11.0886

LIMIT=100

7=0

CALL FUNCT(FUNCT,X,C,CP,EST,ERR,LIMIT,IR,VC)

WRITE(6,201) X(1),Y(1)

200 STOP IF(X,IR,IR=1,IR)

WRITE(6,101)IR

101 STOP IF(X,IR=1,IR/XY,IC=1,512.5)

WRITE(6,202) X

400 STOP IF(X,IR,IR=1,IR/XY,IC=1,512.5)

WRITE(6,102) X(1),Y(1),I=1,12

102 STOP IF(X,IR(1,13,1)=1,512.5)

WRITE(6,201)

201 STOP IF THE DIGITAL FILTER HAD THE FORM $Y(Z) = \frac{Z^2 + A(I)Z + B(I)}{Z^2 + C(I)Z + D(I)}$, $T = 1$

DO 107 I=1,

Y(I)=CABS(ARSH(I))

107 PHASE(I)=ATAN(AMAG(H(I))/REAL(H(I)))

AI=1.0

```

K1=R+1
K2=2*K+1
K3=3*K+1
A(I)=(-2.0+2.0*X(K1)**2)/(1.0+X(I)**2+Y(K1)**2)
C(I)=(1.0-X(I)**2+X(K1)**2)/(1.0+X(I)**2+Y(K1)**2)
C(I)=(-2.0+2.0*X(K2)**2)/(1.0+X(I)**2+Y(K2)**2)
C(I)=(1.0-X(I)**2+X(K2)**2)/(1.0+X(I)**2+Y(K2)**2)
F1=C(I)*PLX(C(I)**2-X(I)**2,0.0)
F2=C(I)*PLX(A(I)**2-4.0*X(I),0.0)
POLF1(I)=-C(I)/2.0+CS*F(I)/2.0
POLF2(I)=-C(I)/2.0-CS*F(I)/2.0
ZPOL1(I)=-C(I)/2.0+CS*F(I)/2.0
ZPOL2(I)=-C(I)/2.0-CS*F(I)/2.0
A)=1+(1.0+X(I)**2+Y(K1)**2)/(1.0+X(K2)**2+Y(K3)**2)
WRITE(6,105)I,A(I),I,(I),I,C(I),I,F(I)
105  F=0.0;A(I),I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0;C(I),I,
I)=1.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0
WRITE(6,106)I,I,L=1(I),I,POLF1(I),I,ZPOL1(I),I,ZPOL2(I)
106  F=0.0;A(I),I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;
F12.5/50.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;
I)=1.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;F12.5/50.0;I,(I),I)=1.0;
104  C(I)=0.0
DO=100.0
WRITE(6,100)I
100  F=0.0;A(I),I,(I),I)=1.0;F12.5/50.0
WRITE(6,100)I
100  F=0.0;A(I),I,(I),I)=1.0;F12.5/50.0
CALL PLT3(6,Y,I)
110  F=0.0;A(I),I,(I),I)=1.0;F12.5/50.0
DO=100.0
I=101
POLF1(I)=REAL(POLF1(I))
POLF2(I)=REAL(POLF2(I))
POLF1(I)=ALAS(POLF1(I))
POLF2(I)=ALAS(POLF2(I))
ZPOL1(I)=REAL(ZPOL1(I))
ZPOL2(I)=REAL(ZPOL2(I))
ZPOL1(I)=ALAS(ZPOL1(I))
ZPOL2(I)=ALAS(ZPOL2(I))
CALL PRINT(6,I,I,I,I,ZPOL1,ZPOL2)
N=101
X(I)=X(4)
Y(I)=Y(4)
X(I)=X(7)
Y(I)=Y(7)
X(I)=X(2)
Y(I)=Y(2)
X(I)=X(1)
Y(I)=Y(1)
GO TO 100
STOP
END

```



```

SUBROUTINE FUNCT(N,X,Q,DO)
DIMENSION X(N),DO(J),YD(20),S(20)
COMMON/7(EN),S(2,20),Y(2,20),X(20),Z,1,2,3,4,CPLX
COMMON/R1/Z,H
COMMON/R3/AG,YD
COMMON/R4/K,K
Q=0.0
SHT1=0.0
SHT2=0.0
DO 2 J=1,N
P=CMPLX(1.0,0.0)
DO 1 I=1,K
K1=K+I
K2=2*K+I
K3=3*K+I
RY(I,J)=(Z(J)**2)*(1.0+(X(I)**2)+(Y(K1)**2))+Z(J)*(2.0*(Y(K1)-
1.0)-2.0)+(1.0-(X(I)**2)+(Y(K1)**2))
R2(I,J)=(Z(J)**2)*(1.0+(Y(K2)**2)+(Y(K3)**2))+Z(J)*(2.0*(
1.0 X(K3)**2)-2.0)+(1.0-(Y(K2)**2)+(Y(K3)**2))
1 R=RR(I,J)/RY(I,J)
H(J)=P
SHT1=SHT1+COS(H(J))*YD(J)
2 SHT2=SHT2+COS(H(J))*YD(J)
ZC=SH1/SH2
DO 3 J=1,N
F(J)=1.0-COS(H(J))-YD(J)
3 F=1.0-F(J)**2
DO 4 I=1,K
K1=K+I
K2=2*K+I
K3=3*K+I
S1=0.
S2=0.
S3=0.
S4=0.
DO 5 J=1,N
PL=2.0*F(J)*H(J)+COS(H(J))
1 F=(2.0*Y(I)*(F(J)**2)-2.0*Y(I))/F(I,J)
A2=2.0*Y(I)*(F(J)**2)+4.0*Y(I)*Z(J)+4.0*Y(I))/F(I,J)
2 F=-(2.0*Y(I2)*(F(J)**2)-2.0*Y(I2))/F(I,J)
A4=-(2.0*Y(I3)*(F(J)**2)+4.0*Y(I3)*Z(J)+2.0*Y(I3))/F(I,J)
S1=S1+PL*FAL(I1)
S2=S2+PL*FAL(I2)
S3=S3+PL*FAL(I3)
5 S4=S4+PL*FAL(I4)
DO(I)=S1
DO(K1)=S2
DO(K2)=S3
4 DO(K3)=S4
RETURN
END

```

VITA AUCTORIS

- 1945 Born on the 24th of March in Alexandria, Egypt.
- 1963 Matriculated from Victoria College , Alexandria, Egypt.
- 1968 Graduated from the University of Alexandria with a B.Sc. degree in Electrical Engineering.
- 1970 Registered as a Candidate for M.A.Sc. in Electrical Engineering at the University of Windsor.